(continued from part 28)

Viewdata

Videotex is the term that covers both teletext and viewdata. **Teletext** describes the service provided by the BBC (Ceefax) and by ITV (Oracle) and involves the transmission of information to suitably equipped television sets.

Viewdata involves the transmission of information either to television receivers, or to viewdata-only terminals, from a computer via telephone lines. The most well known viewdata service is Prestel which had a disappointing start in the early 1980s. Prestel's target is now one million users by the end of 1985. Prestel operates in a similar way to teletext: it is only available to those with the correct equipment.

However, Prestel can also offer private facilities as well as universal information access. This is achieved through **closed user groups** (CUG) which allow a subscriber to access information on the viewdata base providing they know the password. Companies can run a private viewdata system on the back of Prestel.

Below: video teleconferencing – groups of individuals at separate locations can communicate by sight and sound.

The Research House/British Telecom

The Prestel subscriber can also transmit data. The system is 'interactive' and using a viewdata keyboard, or pad, users can order goods and pay for them by keying in a credit card number.

There are more private viewdata systems in Britain than in the rest of Europe. Germany, though, is close behind, and France is installing small, basic VDUs at little extra cost in an effort to build the biggest computer network in Europe (about 300,000 installations are planned).

Many of the benefits of viewdata are not characteristics of the host system, but of the networking technology and the design behind it. It is the choice of network that determines standardisation of the user interface. It also decides the speed and efficiency of the system, as well as its cost of implementation and use.

A videotex service is a two-way information flow by way of a telephone, a television and a typewriter keyboard. Videotex is therefore a communications medium, not a database access method, and this is its true strength.

Videotex can create market opportunities in information industries which would otherwise be impractical or impossible. It can extend markets geographically, or in terms of the level at which a product is pitched. It can also reduce costs of communications or information storage (paperwork, mail, telephones, staff, transport and equipment) by speeding operations and by improving cash flow and efficiency. Customer services can be improved due to improved dissemination of central facilities, thereby increasing the capacity of a business by automating many time-consuming functions. This is of particular benefit to businesses with seasonal peaks of demand, such as travel agents.

Voice processing

A system integrator can implement voice technology into office systems in a way that allows migration from a single user system, to one that encompasses both company internal use and external use via communications networks.

Human beings communicate with computer terminals in ways that usually require special training or programming

skills. Voice technology addresses the need for a more human man/machine interface. The ability to use speech to command the activities of a computer (or a computer-controlled machine) and to have that machine respond with speech output, would make office life very straightforward.

Some industries already use voice processing terminals: banking, retailing and manufacturing, for example.

Users of voice-based transaction systems need very little training and, in addition, the systems are convenient to use. Speech allows a user to enter data faster, and in some cases more accurately. It also avoids the need for transcription of voice data and any intermediate processing steps that this kind of information involves.

The market for speech processing terminals falls into two broad areas: 'busy hands/busy eyes' and remote transaction processing. Busy hands/busy eyes situations are those in which an operator must interrupt an important activity to enter or read other information into a computer terminal. This is particularly so of all the three industries previously mentioned, banking, retailing and manufacturing.

Remote transaction processing takes advantage of the telephone, so that a company's customer base or mobile sales force can interact with a centralised information system. An example is the reservations and information system of a major airline or travel agent. The agent uses a computer terminal to query the database and then relays the information to the customer. By having a voice processing terminal answer the telephone, the caller could talk directly with the airline's computer.

At present, voice technology is still fairly primitive. The systems are largely dependent on the speaker, who normally has to train the system either by entering a vocal vocabulary or a set of syllables or phonemes. The system turns these into stored digital data, or digitises them, creating a template of each item in its memory. Spoken words, in turn, are digitised and compared with the templates. If there is a match, the word is recognised. Voice processing is discussed in greater detail in *Digital Electronics 24*.

Electronic filing and retrieval

A fast and accurate method of accessing the data stored in an office system is essential. All the components of the automated office must aid users in their search for information.

Electronic filing and retrieval includes tools for both automatic file searching and automatic retrieval. The system may be either a closed or an open loop. In a **closed loop system**, both the means of location and the information have been electronically encoded. In **open loop systems**, the means of location is electronic, but the information is not.

The purpose of electronic filing is to promote the straightforward management of information, regardless of the recording medium. It follows conventional file systems precepts.

According to various surveys, about 35% of all filed papers are never retrieved and up to 95% are never accessed after the first year of storage. Another interesting figure is that 1-5% of all documents are misfiled – the average cost of a misfiled document is between £20 and £35. (These figures are taken from "Survey of Tools and Technology" by David Barcomb – produced in conjunction with Digital Equipment.)

Electronic filing and retrieval has many advantages, for example:
1) faster access to information;
2) reduction in misfiling;
3) reduction in amount of office floorspace;
4) storage efficiency through shared access;
5) portability of files, e.g. floppy disks;
6) limited dependence of human knowledge of filing techniques.

The only limitations to electronic filing are that the user must have access to a terminal and that it takes a few seconds to dial-up a computer system and log on unless the user is already on-line.

Use of a computer-based message system

In most message systems, the text of every message appears only once, regardless of the number of persons to whom it is addressed. The message itself does not

appear in the files of the originator, instead, one master file maintains all the messages. All that exists in the originator's files (and the files of all subsequent recipients) is the message and a note saying who sent it, the date and time it was sent, the subject and a unique identification of the message.

Such systems also keep and maintain a directory which contains all the names and addresses of system and user-generated files, the number of messages within each file and the size or length of each message.

An example of one method of retrieving filed information, the tree search method, is shown in *figure 2*. The growth of electronic filing and retrieval systems parallels that of electronic databases. Both are vast libraries stored on central computers.

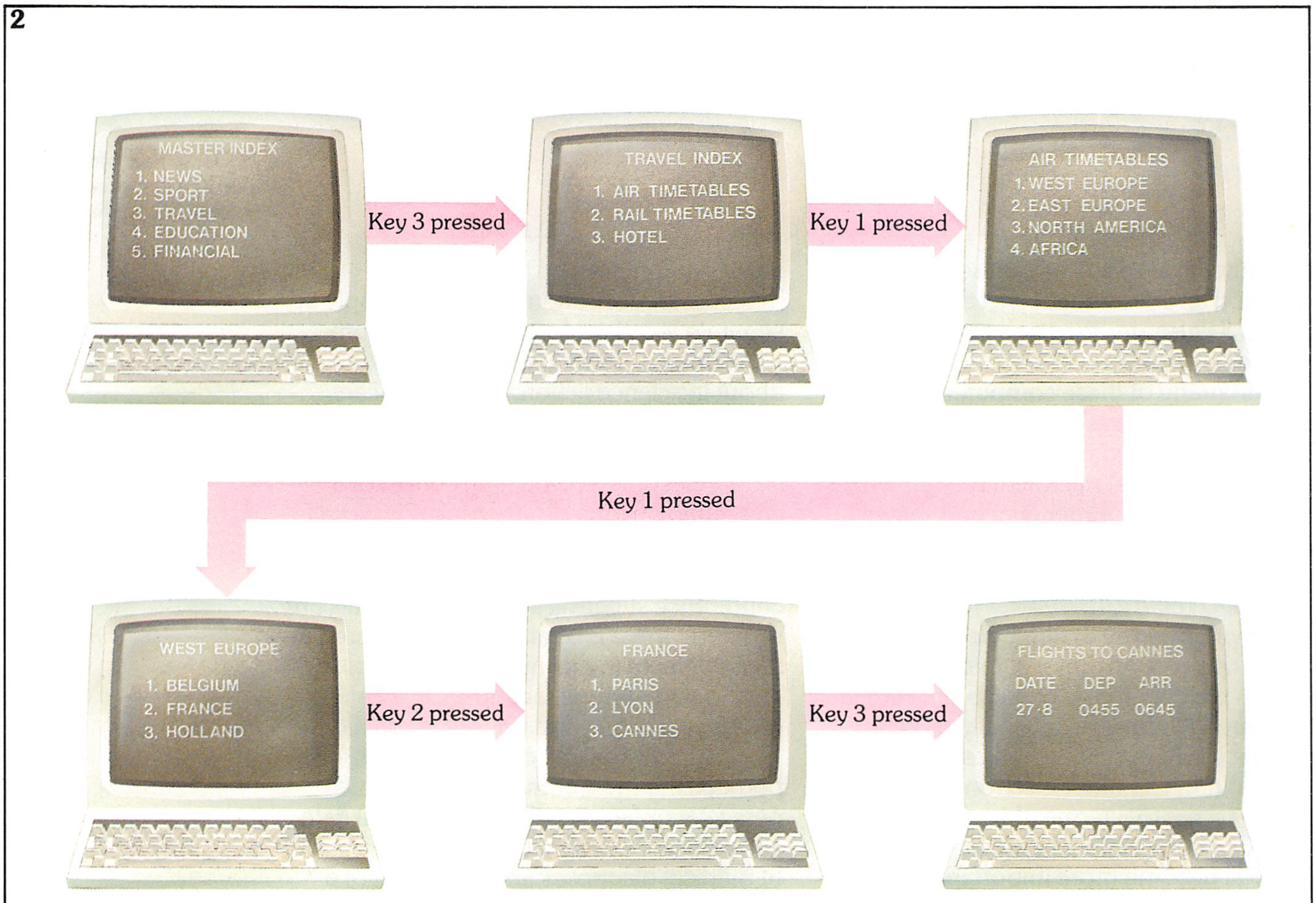
Specialist applications of office automation

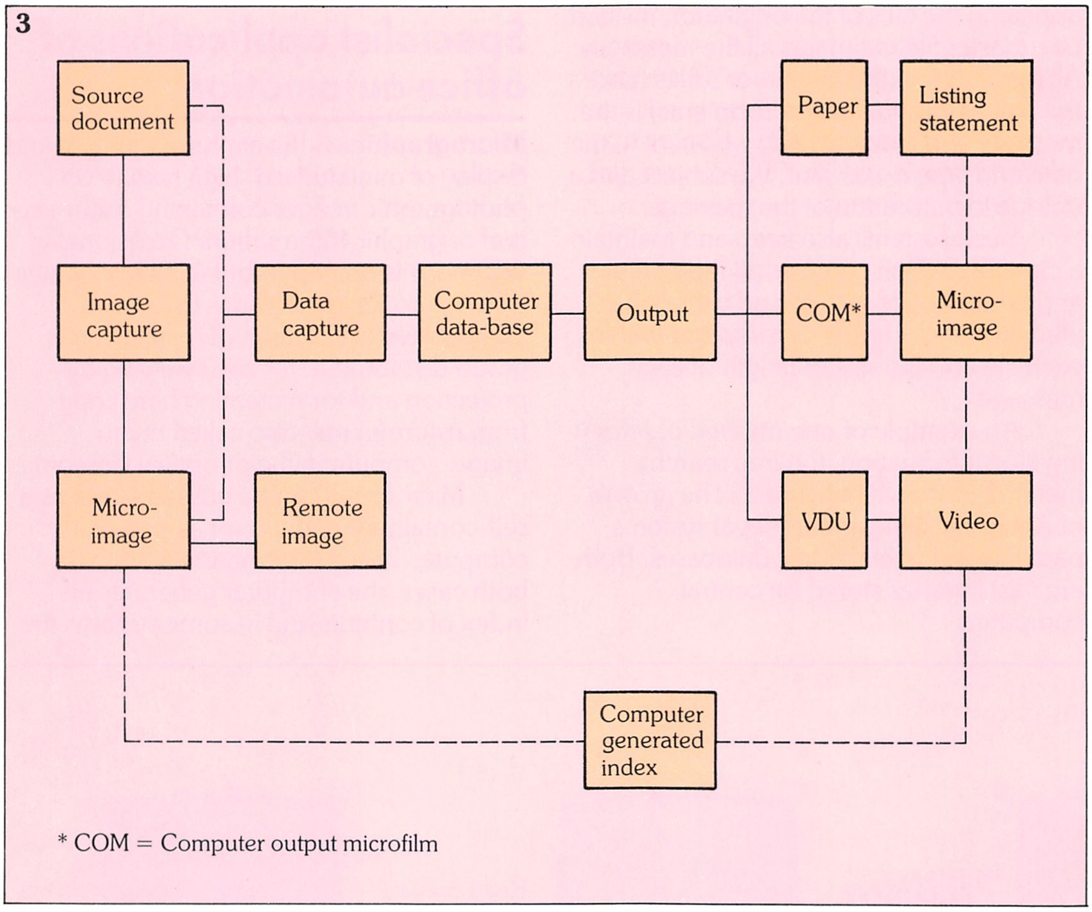
Micrographics is the capture, retrieval and display of miniaturised, high resolution photographic images containing either textual or graphic information. Occasionally, the image is on paper, but usually it comes from microfilm. It provides quick and inexpensive duplication of images for group distribution, for easy viewing by projection and for recreating hard copy from **microforms** (also called micro-image, computer fiche or optical storage).

Microforms can be utilised either as a self-contained medium, or as part of a computer-based information system. In both cases, the computer generates an index of contents and in some systems the

2. The tree search method of retrieving filed information.

2





3. Using microforms as part of a computer-based information system.

computer is used to mechanise roll or fiche selection and presentation to the user. *Figure 3* illustrates the components of such a system.

In many of these computer-based storage and retrieval systems the retrieved images are viewed by the user via a specialised optical microfilm reader, which may also have a printer attached. Later systems, however, incorporate microform output with the ordinary computer output on a VDU (*figure 4*).

With its array of film types, techniques and retrieval options, micrographics affords the user many benefits. The limitations are few: microfilm usually provides no space for annotations (except dust jackets) and like other electronic systems, it needs access to a terminal.

Concurrent filming and database capture will become more important as office use and on-line micrographic databases become more widespread. In the future, there will also be more products that integrate the retrieval of micrographic and alphanumeric, graphic and facsimile databases.

Teleconferencing

Teleconferences are meetings of people at separate locations by means of telecommunications. The idea has been talked about in science fiction novels for years, but only now is the technology becoming generally available to co-ordinate it.

Teleconferencing may consist of a conventional long distance telephone call, or a complex integration of audio, visual and computer elements. Although a teleconference suggests that all participants are separate, there is often a group gathered at some or all of the locations. The teleconference is a mixture of face-to-face, local group interaction and remote interaction.

Administrative conferences are perhaps the most common type of business teleconference. Other important categories include: telemedicine (the conduct of medical diagnoses and treatment of a patient at one location and the doctor or team of specialists at the other); teleteaching; and teleservicing (the long distance diagnosis, service scheduling and service of equipment).

Audio and **video teleconferencing** must take place at the same absolute time (regardless of time zones), but in computer teleconferencing the participants need not be present at the same time. Computer teleconferencing is akin to electronic mail, but the teleconference computer provides more built-in assistance to the users.

Video teleconferencing, with full motion television, is better suited than either audio or computer teleconferencing to discuss complex tangible objects and interpersonal matters. However, video of all types is suitable for only about 8% of face-to-face meetings for which audio alone is insufficient.

The three major advantages of computer teleconferencing are as follows:

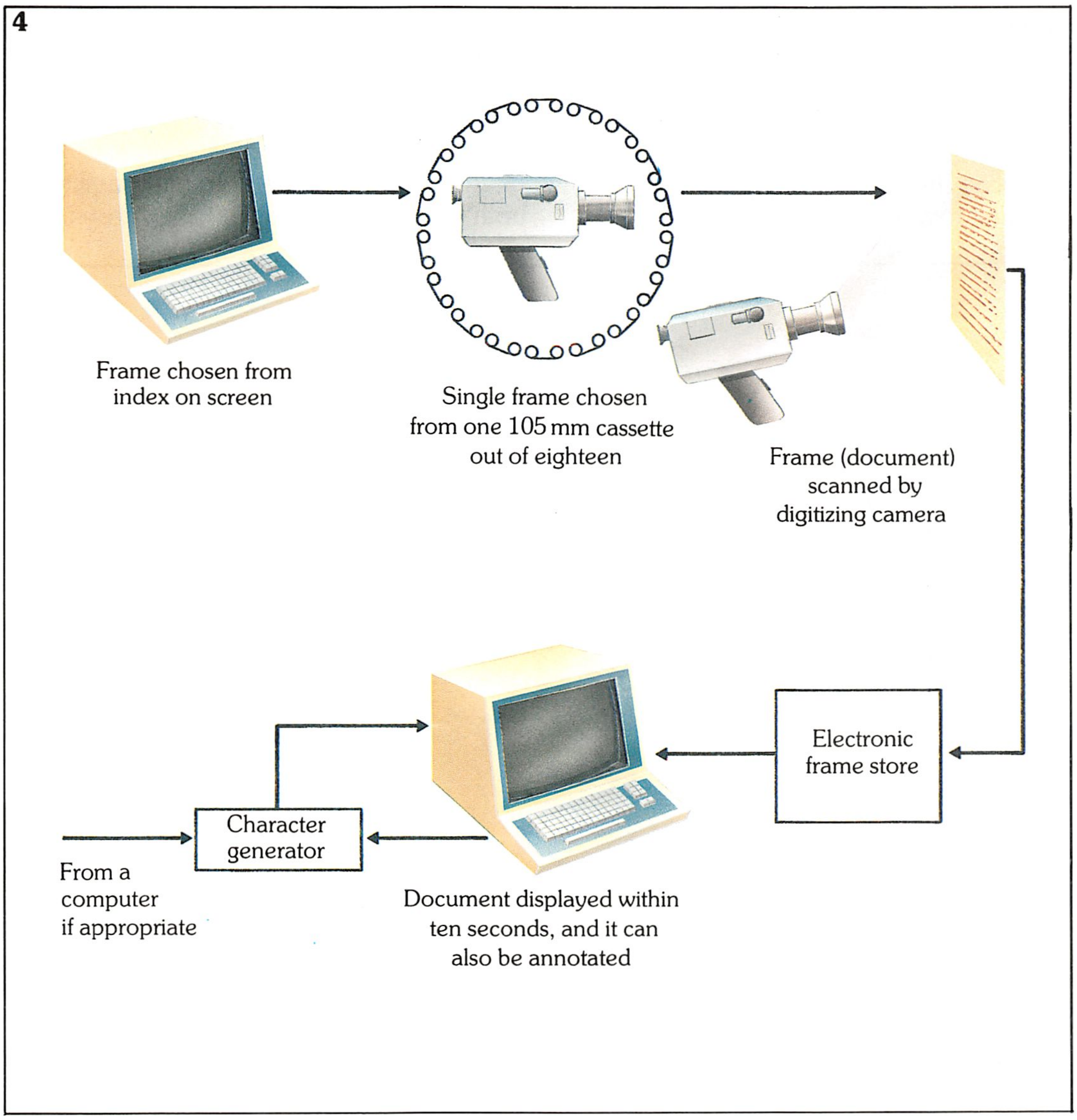
1) time zone transparency;
2) nonconcurrent availability of participants;
3) a written record of proceedings.

The first two advantages are obviously similar: busy personnel within the same building may find it difficult to attend meetings and get together at the same time as American counterparts, for instance.

Unlike other teleconferences, a computer teleconference permits simultaneous comments to be made. Users can enter the teleconference whenever they need to and as often as they need to.

The limited variety of teleconferencing software reflects the fact that very few middle and upper level managerial personnel will key data in using a keyboard.

4. Microform output can now be displayed on a standard computer VDU.



The social implications of office automation

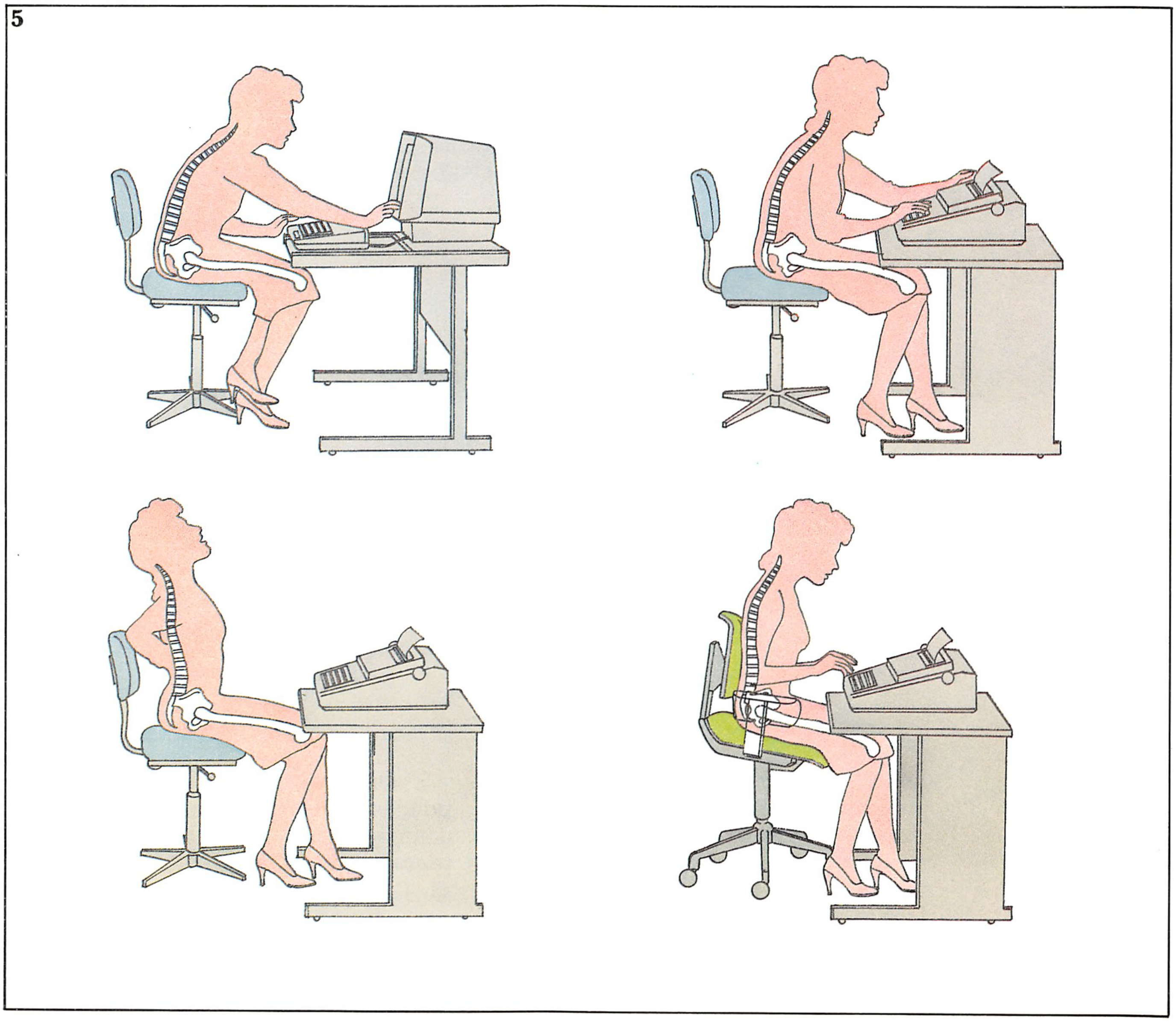
There is little disagreement that technology and office automation will play an ever expanding role in business operations. But management must also recognise that a critical factor in applying such technology, now and in the future, will be the employee's ability and willingness to use it.

If work has to be done in dramatically different ways, then employees are going to be affected significantly. As an example of what can be done, the management of American Insurance company AETNA set up a unit to assess the impact of new technology on its employees. Known as the People/Technology Programs (or P/TP) unit, it has developed programs and policies to address the issues they found most worried its 350,000 employees.

Addressing the human and **ergonomic** factors that arise when people must interact with machines is not an easy task. Ergonomics is the science of studying humans in their working environment, and that is what P/TP set out to do. P/TP's initial strategy was to concentrate on four basic issues of concern: ease of use, health issues, the manager's role, and the data processor's role.

Ease of use pertains to the usability of both hardware and software. An easy or

5. Ergonomically designed office furniture can considerably improve health and comfort, hence increase productivity.



normal-to-use system incorporates the language and procedures that are familiar parts of given business functions. These make a system 'normal' (if not easy to use). The normal-to-use system also recognises that there are some things worth doing on a computer even if they are somewhat difficult.

Health issues include the concern over radiation from computer screens, as well as muscular and skeletal aches that have been associated with using a VDU. Other concerns include the role that lighting and furniture plays in the office environment (*figure 5*). The psychological aspects of stress may also contribute to many physical ailments.

Ergonomics

A normal or easy-to-use system uses the language and procedures that are familiar to workers. The system's screen messages, for example, are more effective when expressed in the language of the customer (an important point). The normal-to-use system may not be the easiest to use, but it should be the most effective for its particular customer.

Many systems are not necessarily easy to learn, and easy-to-learn systems may not necessarily be easy-to-use. For example, an easy-to-use system for experienced employees may involve the use of multiple key operations, or special codes to make things quicker. Yet these keys and codes are not standard business operations and must be committed to memory over a period of time.

Alternatively, the system may use software which incorporates extensive menu systems. Menus, while important for the novice, can be cumbersome to the experienced computer user if another way of navigating the system is not provided.

A straightforward office system should include a menu system. When using a menu dialogue, it is important to consider seriously which type of menu is best: fill in the blanks, cursor select, and 'soft key' defined menus are examples.

These are all comparitively easy to use, but some may require fewer keystrokes than others, or may be comfortable for a particular group of users, e.g. managers as opposed to data processors.

Assigning menu defaults that allow the system to suggest the next step in the process is helpful to novices.

The experienced worker should be provided with a command language. This may involve having predefined key words that cause a process to occur when typed in. This allows the user who is more familiar with the menu system to be able to navigate their own way about.

Other alternatives to menu systems include question and answer, query language, natural language (English), or interactive graphics and touch screens.

Careful design of the way in which information appears on the screen can increase a users productivity and reduce input errors. Good screen design should include thoughts on format (arrangement of the data on the screen), content (the subject matter), layout (arrangement of screen content) and style (the way in which data is presented on the screen).

The application of colour is also an important factor in good screen design. In some cases, the use of colour can simplify the use of a system, but care should be taken that the colours are relevant to the operator's task (e.g. highlights important data on enquiry screen) and that it meets the expectations of users (e.g. red for debits or errors). Colour must also be used with consideration for colour blind users, those with tinted glasses and those with monochrome VDUs.

As much as possible, the same thing should happen when you press a particular key. This enables users to transfer their own experiences from one application to another.

Graphics make it simple to tell the system what you are doing. Using a set of symbols, or **icons**, can make things easy for occasional users, especially management. Any symbols that are used should be familiar to the user and not totally new.

The operator should not have to fill the screen with data that the computer already has. For example, given a policy number, or such like, the system should be able to determine the name of the policyholder and any other constant data when it has this information available.

Using a mouse, a joystick or a touch sensitive screen can reduce error rates and

Left: electronic mail terminal – BCD TeleMail TM120. (Photo: BCD).

increase productivity in some situations. However, the long term values of such tools are not yet determined. When there is need for minimal use of the keyboard (occasional users and management that refrain from typing), such equipment can be faster than a typed command.

The touch screen appears to be the most natural and convenient of the three. It allows the most direct hand/eye interaction and does not require additional desk space, or other devices.

However, it is much less exact: the width of the finger is quite large compared with the resolution of the graphics display. Therefore, it cannot be used for anything very detailed, e.g. drawings, but is reserved for indicating choice between alternatives (as on a menu).

The systems that we have been talking of also help to minimise the stress and operator fatigue that occur with the extended use of VDUs. This may be in the form of natural breaks where the screen reports to the user 'task completed'. Clear and consistent presentation of data decreases the number of eye movements that an operator may make, thus reducing the chances of eye strain. Errors result from eye strain and a general fatigue ensues.

Light in an office is important. It can be difficult to see the screen under some office lighting. It is reported that fluorescent lighting is notorious in causing eye strain and headaches. To help prevent some of these complaints manufacturers have provided gauzes and polaroid filters that fit in front of the screen.

Although VDU screens have improved in quality enormously in the last few years, it's not the machine that's the problem – it's the way that the machines are used, and the way people work at them constantly. EEC regulations now recommend breaks every ten minutes to prevent eye strain and fatigue.

Consultation and sensitive management are important over this issue. This change must be made gradually with a positive reasoning and organisation behind it. This also helps management to become aware of the new technology and eventually to be able to use it themselves.

Role of data processors

The traditional data processor's role will change in the office of the future. The proliferation of the microcomputer and the growing computer sophistication of end users has led to applications developments in areas other than systems. As users take on more responsibility, professional data processing programmers and integrators are assuming more the role of consultants.

This does not mean that their jobs are any less skilled, but they should realise that even more users are becoming computer literate.

The manager's role

What are the personnel issues involved with using new technology, and how do they affect the working environment? Well, the manager's role is essential in handling personnel issues effectively, but before the manager can attempt to find a solution, he or she must understand the problem, which is no easy task.

It takes time to sort good information from misinformation. Most technology-oriented literature, like advertisements, tell us that there's nothing technology can't do. But the popular press and the most recent ergonomic reports tell us otherwise.

The opinions of those using new technology today fall into two camps: those with problems and those without problems. Employees without problems tend to be those that had participated to some degree in decisions to automate. These are the technical people, the data processors. Others who have welcomed the change are secretaries who now have word processors. They can now cope with managers changing their minds over documents and letters!

Those employees with problems often had no choice about whether to automate or not. These people were generally happy with the way things were.

On introducing new technology, the error rate of workers tends to rise. Each employee does significantly more work and as a result of shared databases that work is likely to be tied in with other work. Finding and correcting errors can often be more difficult in such situations because of the complex and often inflexible ways that some office systems work.

Was it the programmer? Did the operator make an error? Did the manager fail to provide adequate training? It is easy to see how the phrase 'the computer did it' became popular.

The main reason for many of the complications brought about by technology rests in the nature of office work itself. People and machines have very different needs in regard to the structure of work.

For machines, work needs to be broken down into very small parts, with little variation. However, people prefer to have more challenging work. They like variation. It gives them the opportunity to learn new skills and to avoid boring repetitive tasks. Perhaps most importantly, they want the freedom to adopt a working style that suits them best.

The business manager must balance the needs of both people and technology. The concerns of those employees who don't feel managers are handling technology well are cause for close attention. What are these concerns?

Common fears

Employees fear that technology will replace them. This fear is very real, although the facts in some cases do not justify it. In many cases, new technology produces additional jobs and offers opportunities to develop new skills.

However, some of the less skilled personnel have some cause to worry. If the people that spend each day updating a huge card system of names and addresses cannot find a computerised role within a company, they may find themselves replaced by a database management system and a micro.

Fear of complexity is a more common concern. Many people are afraid of failure and are naturally concerned that they will not be able to operate the new equipment. They fear that their peers, who can use the technology, will lose respect for them. This of course highlights the reason for adequate training and obvious management involvement.

Workers are confused by the jargon that inevitably accompanies new equipment and they must learn a whole new vocabulary to be able even to talk about a system.

All these concerns can distract people from being productive. So what is the answer? People are the foundation of any business and people are always more important than machines. Therefore most of the problems that arise from the introduction of technology relate to good management practice.

In the electronic office of the future, the successful manager will be able to recognise and address the important issues before they can affect the work environment. The result will be an office where people can effectively apply the power of office technology to accomplish their work.

COMMUNICATIONS

The telephone-2

Signals and multiplexing

In *Communications 2* we saw how telephone terminals using loop-disconnect and dual tone multifrequency signalling can form a telephone system with a network of exchanges. Communication links between exchanges are generally over trunk routes, and the type of signalling over these trunks depends largely on the media used. We can now look more closely at the various signalling methods available in a large, modern telephone network.

Four wire operation

The examples of telephone terminals and local exchanges we have seen, use only two wires to convey speech communications in both directions – **two-wire operation** – and are summarised in *figure 1a*. Over short distances, say, from terminal to local exchange (no more than 1.8 km), this signalling method is reasonably economical, because circuit losses due to attenuation etc. are low over such distances and repeaters are not required. Over longer distances of twisted-pair wires, however, losses become significant and correction for losses therefore becomes necessary.

Figure 1b shows how two-wire lines may be used over trunks, with the required circuits forming repeaters. The effect of each repeater in the line is to act as a two-way amplifier, enlarging speech signals in both directions to the required amounts, simultaneously. A simple amplifier of this form is impossible to build, and so the two-way signal has first to be split into separate signals in each direction known as **go** and **return** signals. Each signal is then amplified to the required size before recombination. The device which allows this splitting/recombination of signals is known as a hybrid transformer and is shown in greater detail in *figure 2a*. Its circuit symbol is shown in *figure 2b*.

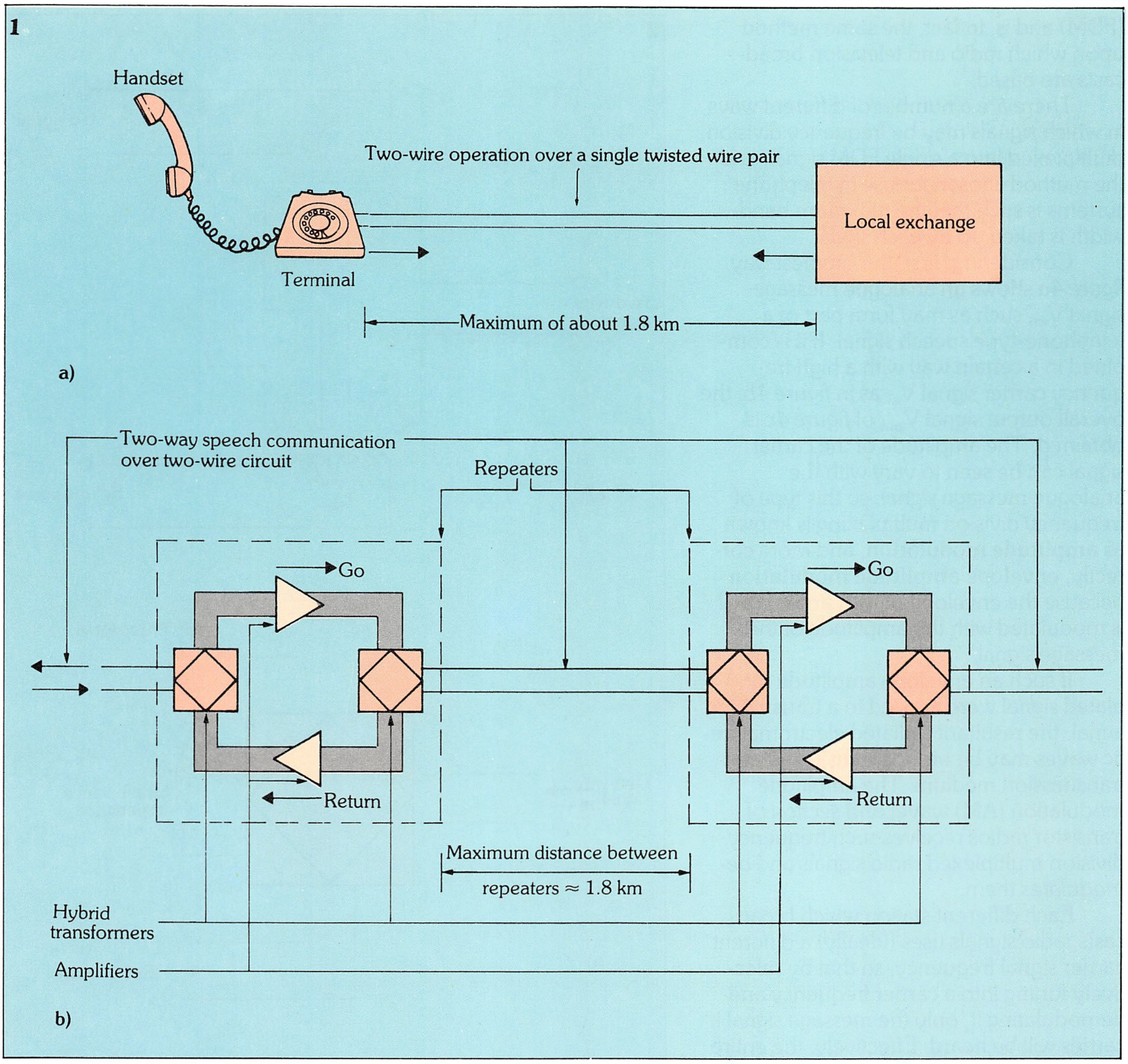
Consider the hybrid transformer's operation. Coils L_2, L_3, L_4 and L_5 are identical. A return signal, v_R, from a four-wire link is applied across coil L_6 and voltages of $v_R/2$ are induced across coils L_4 and L_5. This causes currents to flow through L_2 and the two-wire line, and also through the line balance impedance and L_3. Coil L_3, however, is connected in a reverse way to coil L_2, and so the fluxes produced in coils L_2 and L_3 are in antiphase and so will cancel each other out. The voltage induced across coil L_1 will therefore be zero.

Coils L_2 and L_4, on the other hand, are in phase and in series, so the two-wire output voltage will be equal to v_R.

Impedance Z_o is chosen to be the same value as the impedance of the two-wire line and so, theoretically, no cross-over between go and return lines occurs. In practice though, some cross-over does occur which, if too much amplification is included in the four-wire repeaters, can cause oscillation (sometimes known as **singing**). So correction for losses is limited by the hybrid transformer's line-matching performance and, generally, each repeater/repeater combination, including the losses in the trunk line, and the gain of the amplifiers, must cause an overall *loss* of about 3 dB to prevent oscillation.

Four-wire operating circuits, although using twice as much wire, are simpler to accomplish and use. *Figure 3* shows a possible four-wire circuit and we can see that only two hybrid transformers are required – one at each end of the line – and that the repeater stages comprise simple amplifiers. The loss required to prevent singing (known as the **singing margin**) is now only 3 dB *over the whole line*, and not 3 dB per repeater/repeater combination as occurs in two-wire operation.

1



1. (a) Two-wire; and (b) four-wire operation of telephone terminals and local exchanges.

Multiplexing signals

As we have seen, four-wire circuits of this form are possible, and some do exist. However, they are wasteful because two twisted pairs of wires are necessary for each communications link between two telephone terminals. It is far more economical to multiplex *many* telephone signals together and transmit them along a single transmission medium – generally coaxial cable, although fibre optic cable is now replacing coaxial cable over some major trunk routes. In this way, trunk circuits between exchanges may effectively be *shared* by many signals – this helps to reduce cable laying and materials costs.

Two main methods of multiplexing signals exist in the telephone network – the method chosen usually depending primarily upon the exchanges at each end of the trunk.

The first we shall consider is an analogue method in which the telephone signals are modulated onto specific **carrier signals** at the transmission end of a trunk. Demodulation of these signals occurs at the trunk's receiving end. This method is known as **frequency division multiplexing**

(FDM) and is, in fact, the same method upon which radio and television broadcasts are based.

There are a number of different ways in which signals may be frequency division multiplexed into a single FDM signal, and the method chosen for use in telephone systems is such that the minimum bandwidth is taken up by each signal.

Considering, first, the simplest way, *figure 4a* shows an analogue message signal V_{in}, such as may form part of a telephone-type speech signal. If it is combined in a certain way with a high frequency carrier signal V_c, as in *figure 4b*, the overall output signal V_{out} of *figure 4c* is obtained. The amplitude of the carrier signal can be seen to vary with the analogue message signal, so this type of frequency division multiplexing is known as **amplitude modulation**, and more correctly, **envelope amplitude modulation** – because the envelope of the carrier signal is modulated with the amplitude of the message signal.

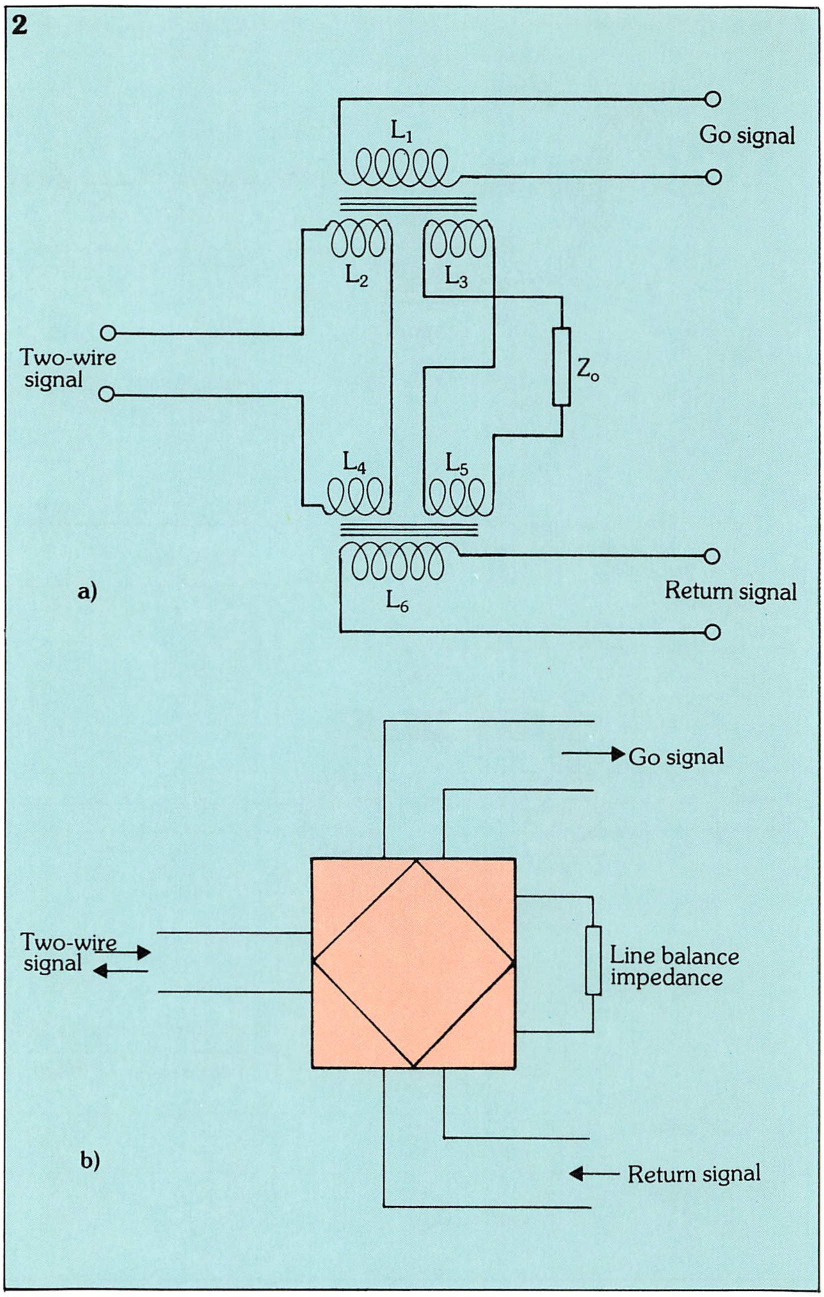
If such an envelope amplitude modulated signal were applied to a transmitting aerial, the resultant radiated electromagnetic waves may be broadcast using air as a transmission medium. The amplitude modulation (AM) waveband section of transistor radios receives such frequency division multiplexed radio signals and *demodulates* them.

Each different station which broadcasts radio signals uses (ideally) a different carrier signal frequency, so that by selectively tuning into a carrier frequency and demodulating it, only the message signal it carries will be heard. Effectively, the entire electromagnetic wave frequency range is divided up into allotted portions – one for each radio station – hence the name frequency division multiplex.

If we look at graphs of signal power against frequency (known as **spectrums**) caused by frequency division multiplexed signals, we may reach some important conclusions. *Figure 5* shows a representation of such a spectrum and we can see that the graph consists of a number of parts which are, in fact, common to any envelope amplitude modulated signal.

This spectrum includes a component at the carrier frequency f_c. Finally, on either side of the carrier frequency component are transposed versions of the original frequency components of the message signal. The **upper sideband** consists of a direct transposition of the message signal band, but the **lower sideband** is a mirror image of the message signal band.

In an AM radio system, the message band is filtered out and only the carrier signal, complete with upper and lower sidebands, is transmitted. *Figure 6* illustrates a possible spectrum for a number of AM radio signals, each with its own carrier. It is transmitted radio signals such as these



2. (a) A hybrid transformer splits each signal into a go and return; **(b)** circuit symbol.

3. Two hybrid transformers are necessary for four-wire operation.

4. (a) Analogue message signal; (b) carrier signal; **(c)** AM signal.

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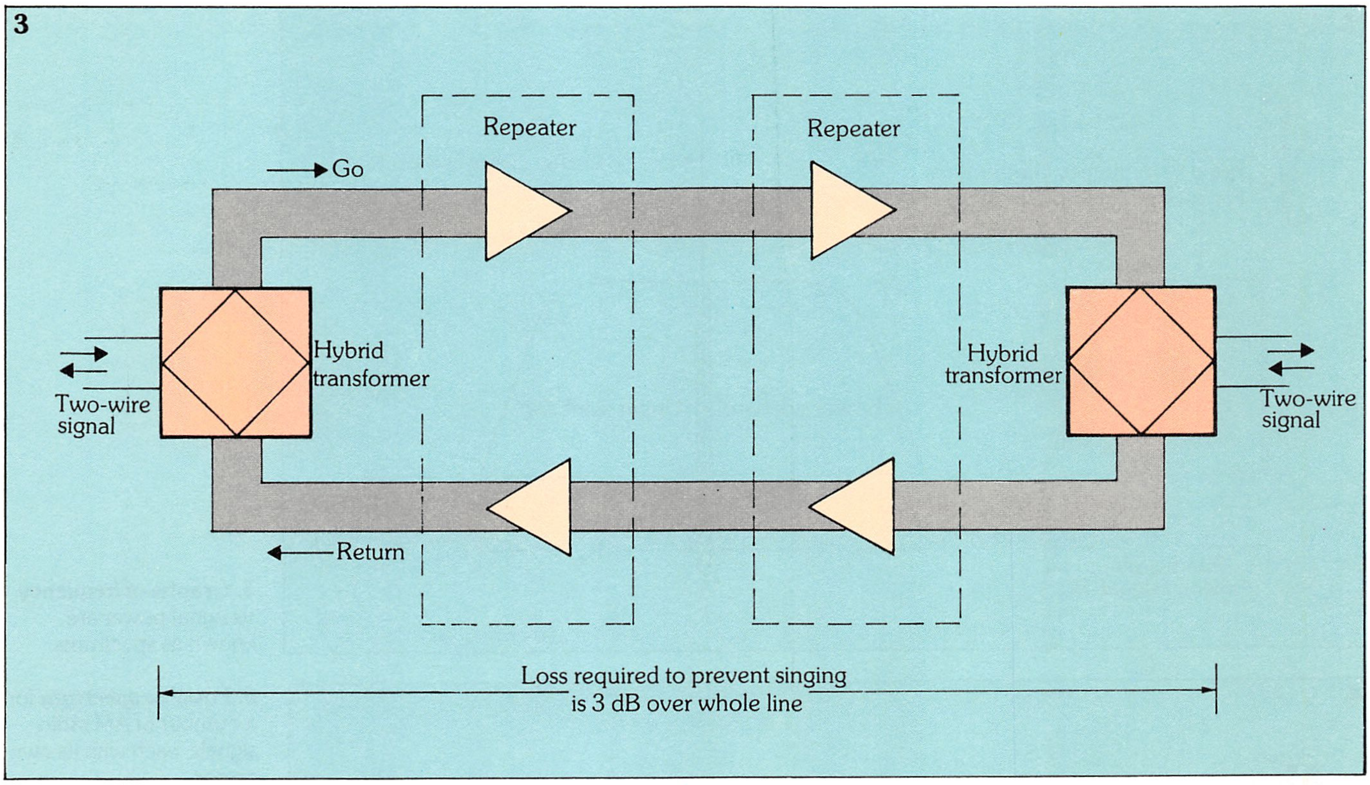

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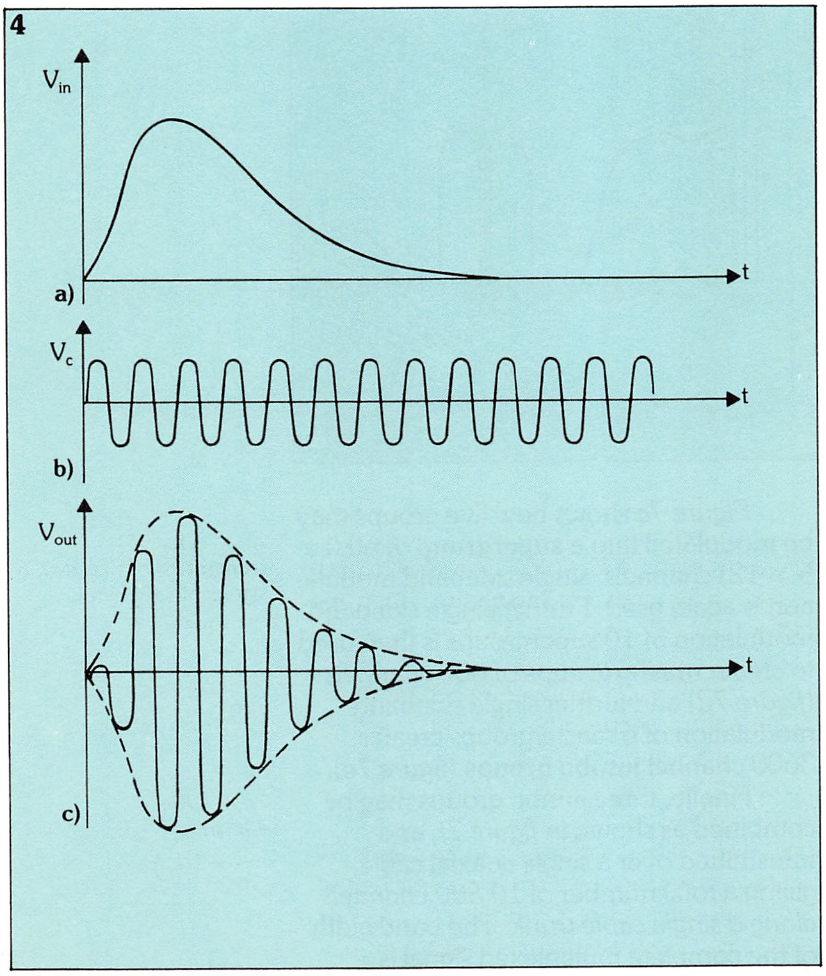

which a radio receiver tunes into and demodulates, to produce the original signal.

The bandwidth of an envelope amplitude modulated signal depends totally on the message signal and is $2f_{max}$, where f_{max} is the message signal's highest frequency component. So, for a transmitted radio signal of a message signal of 100 Hz to 6 kHz (a typical AM radio signal's message signal), the total AM signal bandwidth is 12 kHz, centred on the carrier frequency.

Telephone signals are modulated in a slightly different way, such that only one of the two sidebands of the modulated signal is transmitted, and no carrier signal, in a process known as **single sideband suppressed carrier modulation**. This obviously reduces (by about half) the bandwidth of the transmitted signal and, in the case of a telephone signal of bandwidth 300 Hz to 3.5 kHz, the total required bandwidth is about 3.2 kHz per transmitted signal.

Frequency division multiplexing of telephone signals occurs in a number of stages. The first stage is straightforward single sideband modulation of 12 telephone channels onto carrier frequencies spaced 4 kHz apart, at frequencies of 64 kHz to 108 kHz. Such a collection of

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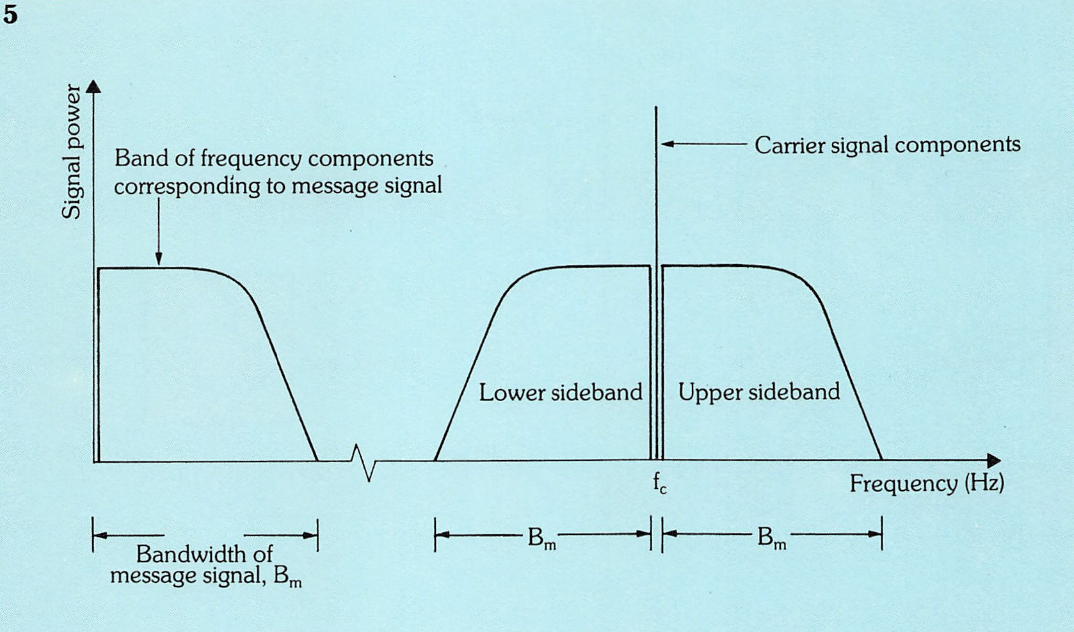

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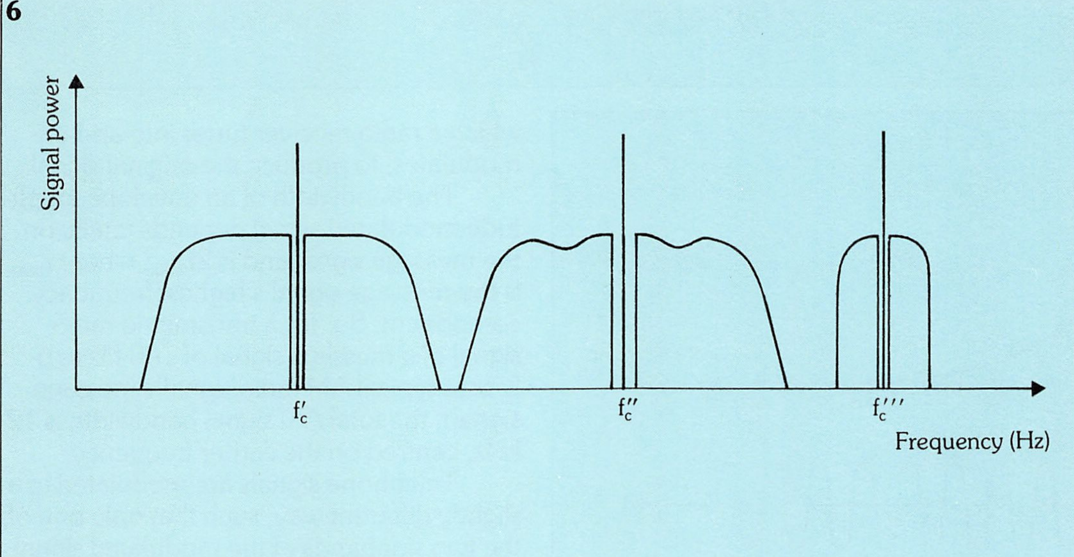


5. Graphs of frequency ***vs*** signal power are known as spectrums.

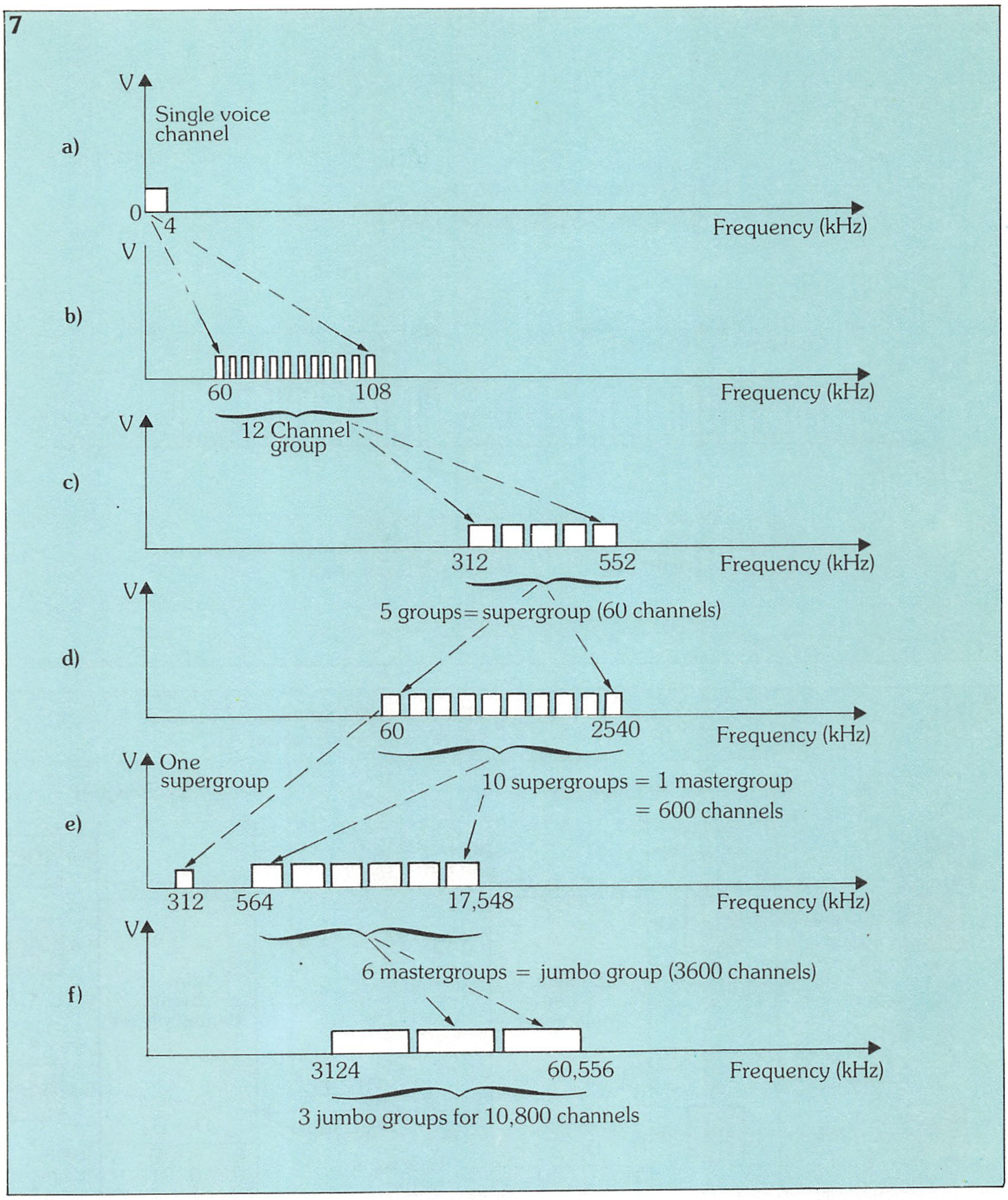
6. Possible spectrum for a number of AM radio signals, each with its own carrier.

telephone channels is known as a **group**. The lower sidebands are used and a typical spectrum is shown in *figure 7b*.

Such a 12-channel group could be transmitted along a single pair of wires, allowing 12 one-way speech signals to share the same cable without interfering with one other. However, there is no need to stop here. Most wires can pass components of much higher frequencies than 108 kHz, so the process can continue – with single sideband modulation of these already modulated groups, so that even more telephone channels can be transmitted over a single transmission link.

Figure 7c shows how five groups may be modulated into a **supergroup** of 60 (i.e 5 × 12) channels; single sideband modulation is again used. Further single sideband modulation of 10 supergroups is then used to create **mastergroups** of 600 channels (*figure 7d*) and further single sideband modulation of 6 mastergroups creates 3600 channel **jumbo groups** (*figure 7e*).

Finally, three jumbo groups may be combined as shown in *figure 7f*, and transmitted over a single coaxial cable, giving a total number of *10,800 channels along a single cable trunk*. The bandwidth of the complete multiplexed signal is



7. Frequency division multiplexing.

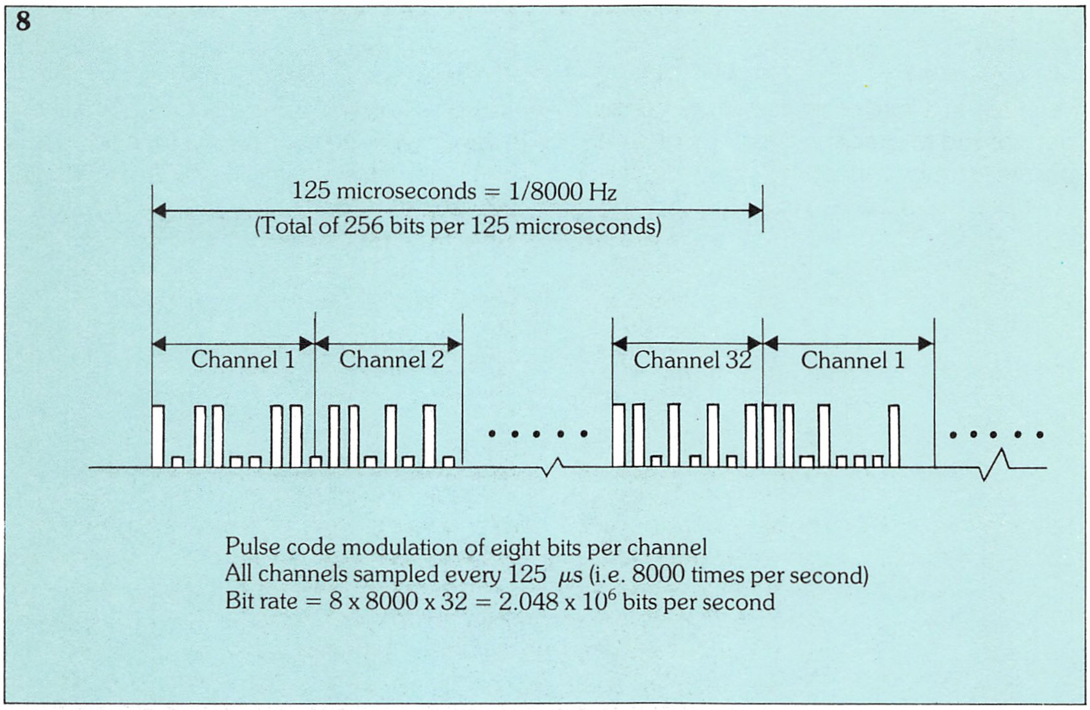
approximately 60 MHz.

Each exchange involved in such FDM links must have the modulation and demodulation circuits to assemble and dissemble the groups, supergroups, mastergroups and jumbo groups, and this obviously increases the cost of the exchanges. Offsetting this, however, is the tremendous capacity and saving in wire and line costs between exchanges. Also, all of these FDM techniques may be used, not only with coaxial cable as the transmission medium, but with microwave, radio, satellite and even fibre optic links.

Time division multiplexing

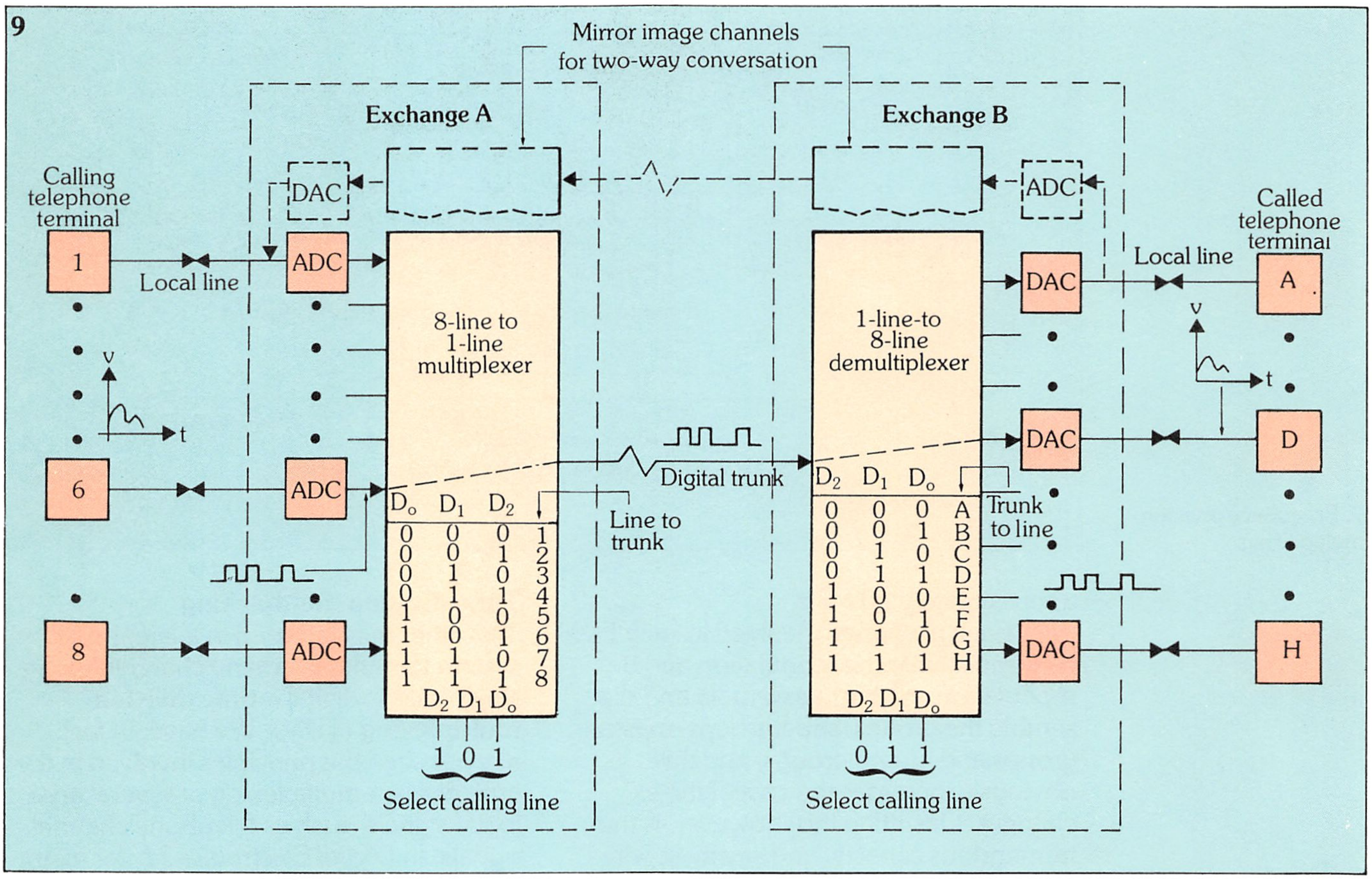
The other method used in a telephone system to multiplex many channels along single links is digital – **time division multiplexing** (TDM). We have, in fact, already seen the principles involved in the time division multiplexing of several analogue signals, such as telephone channel signals. In *Digital Electronics 21*, for example, we saw how many signals may be multiplexed into one by sampling each signal in turn and combining each sample into one overall signal.

An analogue-to-digital converter



8. Multiplexing 32 channels onto a single link.

9. Time division multiplexing.



(ADC) may be used to convert each sampled value of the analogue signals into digital codewords in a process known as **pulse code modulation**. In the telephone system, 8-bit digital codewords are used and transmitted serially along the link.

The sampling rate of the analogue telephone channels is defined by the band-

width of the signals. We know from the sampling rule that a signal with a bandwidth from 0 Hz to f_{max} Hz must be sampled at a rate of at least $2f_{max}$ samples per second to prevent aliasing, so telephone signals (with an f_{max} of 3.5 kHz) should be sampled at a rate of approximately 7000 samples per second. In fact, the sampling rate used in the telephone system is 8000 samples per second. The digital bit rate of a single digitally coded analogue signal is therefore 64,000 bits per second (i.e. 8 bits x 8000 samples) along a digital link.

By time division multiplexing many of these pulse code modulated, digitally coded telephone channels, a digital system of telephone channel transmission is built up. For example, 32 channels may be multiplexed onto a single link as shown in *figure 8*, at a bit rate of:

$$8 \times 8000 \times 32 = 2.048 \times 10^6 \text{ bit s}^{-1}$$

Any number of channels may be multiplexed together for transmission between exchanges – the limit depends on the capacity of the transmission medium and on the speed and complexity of the multiplexing and demultiplexing equipment.

Figure 9 illustrates the principle of time division multiplexed pulse code modulated telephone signals, where an analogue telephone channel from telephone terminal 6 is first converted to a digital signal at exchange A, then multiplexed and transmitted over a digital trunk to exchange B. At exchange B, the digital signal is demultiplexed back to an analogue signal which is forwarded to telephone terminal D.

The code used to select the input signal that is multiplexed onto the trunk line is determined and made active by the calling telephone terminal. In this example, where the calling terminal is number 6, exchange A detects number 6 and provides a control code of 101 to the multiplexer, thereby selecting the input signal from terminal number 6 so that the digital signal is transmitted over the trunk to exchange B.

At exchange B, the code to select the local line to terminal D is set by the called telephone terminal number dialled by terminal 6, and so a call is set up from terminal 6 to terminal D.

Exchange switching

So far, in our look at telephone systems, we have only considered complete transmission links between telephone terminals. Only in the last example of a time division multiplexed, pulse code modulated system, have we briefly turned our attention to the actual switching mechanisms and devices which allow a call dialled from one terminal to be routed to another. We know that such telephone systems are called circuit switched, but how does this switching actually take place?

The switching function of an exchange may be shown, in its simplest form, as a **crosspoint array matrix** (*figure 10*) where incoming and outgoing lines are connected in a matrix by crosspoints or switches at every junction. These switches are normally open, so no connections between lines exist in the normal state. Closing a single switch, however, connects an incoming line to a selected outgoing line.

For example, incoming line 3 may be connected to outgoing line E, simply by closing switch SW_{3E}. For convenience, only one matrix is shown for the connection between calling and called terminal, but in practice there is one matrix for each physical wire of the connection – for a two-wire connection there are two matrices operating in parallel, for a four-wire connection there are four matrices, and so on.

Over the years, many different types of mechanisms and devices have been developed to achieve the switching requirements of crosspoint arrays. The majority of these mechanisms are based on the invention of an American undertaker, Almon Strowger, in 1891. **Strowger exchanges** are still in use in the U.K. telephone system. The exchanges are based on a rotating switch **selector**, in which the wiper forms the incoming line **contact** of the switch. As the moving wiper contact rotates around the switch, one step for every pulse of the called terminal number, it makes a connection with fixed contacts at every outgoing line. A motor, or a solenoid, is used to convert the electrical pulse into movement of the selector.

A possible Strowger network is shown in *figure 11* – this forms the

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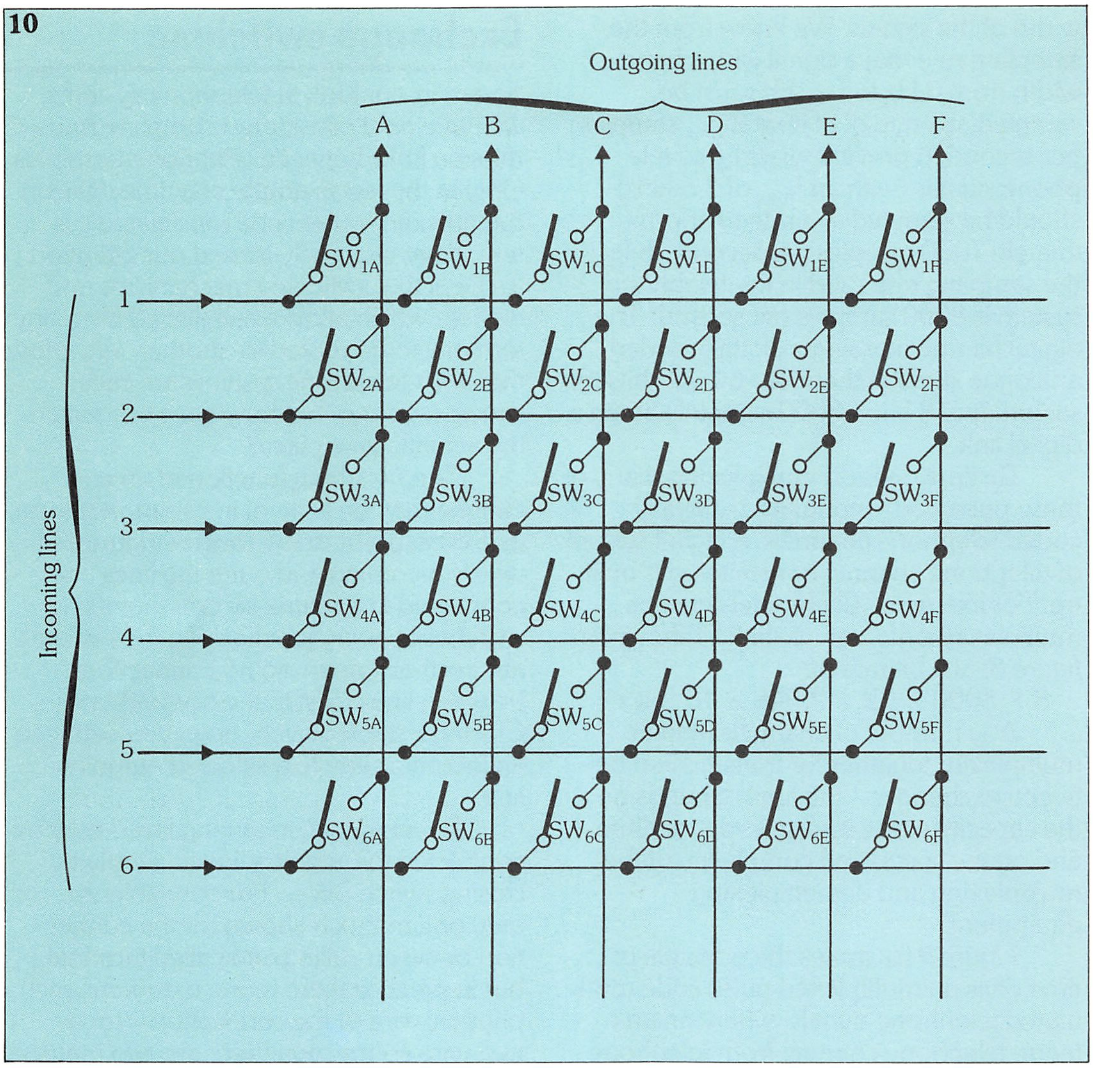


10. Crosspoint array matrix.

equivalent circuit to the six by six crosspoint array matrix of *figure 10*. Any number of incoming lines may connect to any number of outgoing lines by connecting enough of these Strowger selectors in parallel.

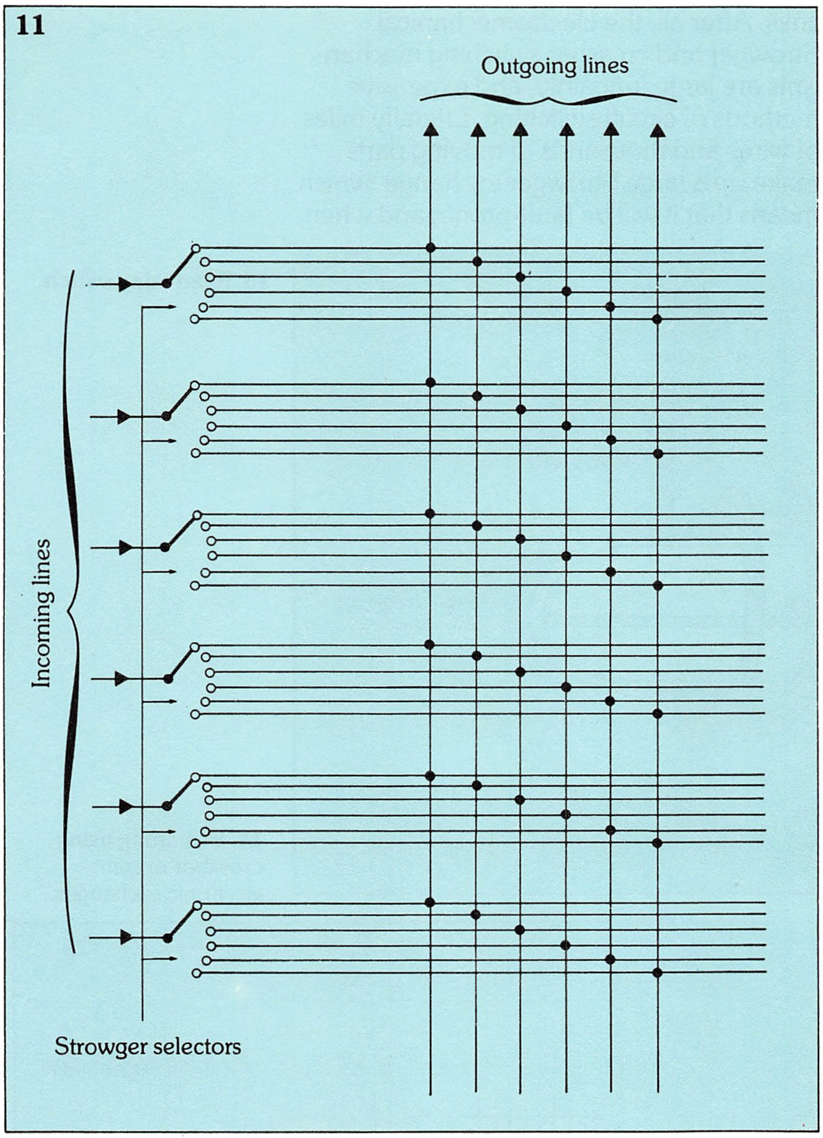
A call between two terminals may be set up using Strowger selectors in a step-by-step fashion, as shown in *figure 12*. As the first digit of the called terminal number is dialled, the wipers of the first selector pair step around, one step for every pulse of the digit. In *figure 12*, the wipers of the first selector pair have stepped around by six contacts.

Similarly, the second, third and fourth digits (and so on) step the following selector pairs around, one after the other as the digits are dialled, until the connection is made between the terminals.

Another main type of electromechanical exchange is known as the **crossbar** system. A crossbar switch mechanism is an array of contacts arranged in rows and columns. Along each row and column is a metal bar, connected to a solenoid, so that as the solenoid operates, the contacts of that row or column move. If the solenoid of a row *and* a column are both operated, the contacts of the crosspoint close.

Semi-electronic exchanges have also been produced in which the crosspoint array is constructed using **reed relay** switches (*figure 13*). Current through a reed relay coil closes the switch and connects an incoming line to an outgoing line.

Figure 14 shows the principle of switching using crossbar or semi-electronic exchanges (although Strowger-type exchanges can be constructed similarly) where the called number is first stored in the exchange before the link is set up. In



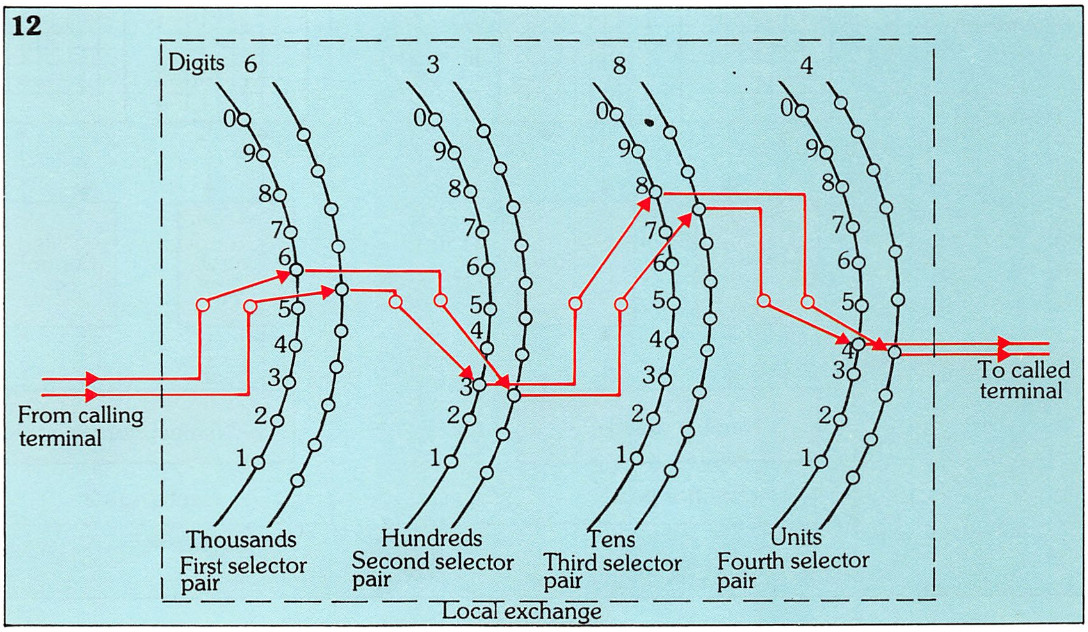
this example, the first two digits of a terminal number define which exchange the called terminal is connected to, and the second two digits define which terminal of that exchange. So, the first two digits, after storage, are used by controls 1 and 2 to define which crosspoint switches are to be closed to make the connection between exchanges.

In the case of terminal A calling terminal B (both of which are connected to the same exchange), the call is routed only through exchange 1, and the second two digits, after storage, are used by controls 1 and 2 to define which crosspoints to close. In the other case, the second two digits are transmitted along the trunk to exchange 2, where they are stored and then used by controls 3 and 4 to close the required crosspoints in that exchange to make the connection between terminals C and D.

The concept of being able to *control* crosspoint arrays rather than directly switching the array from input pulses is an important one, because it means that facilities may be changed or added to a telephone system, simply by changing the controlling equipment and *not* the exchange. If the controlling equipment is a computer of some description, these facilities can be changed even more simply – by *changing the program*. Software control of exchanges is known as **stored program control** (SPC), and is the basis upon which all modern exchanges are built.

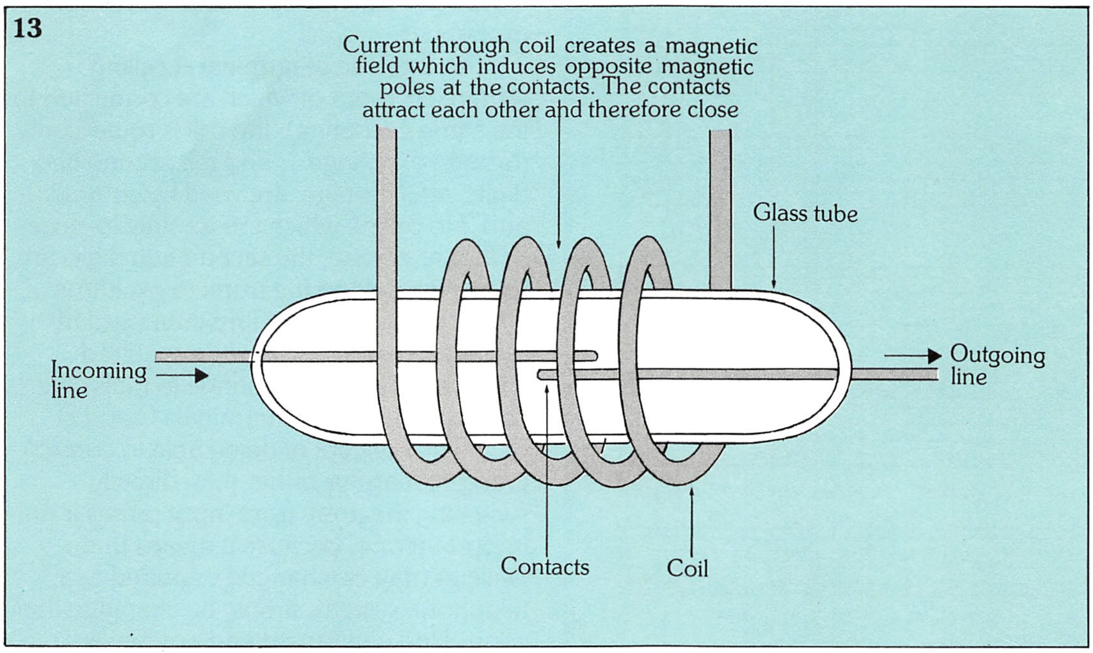
11. Possible Strowger network.

12. Setting up a call between two terminals in a step wise fashion using Strowger selectors.

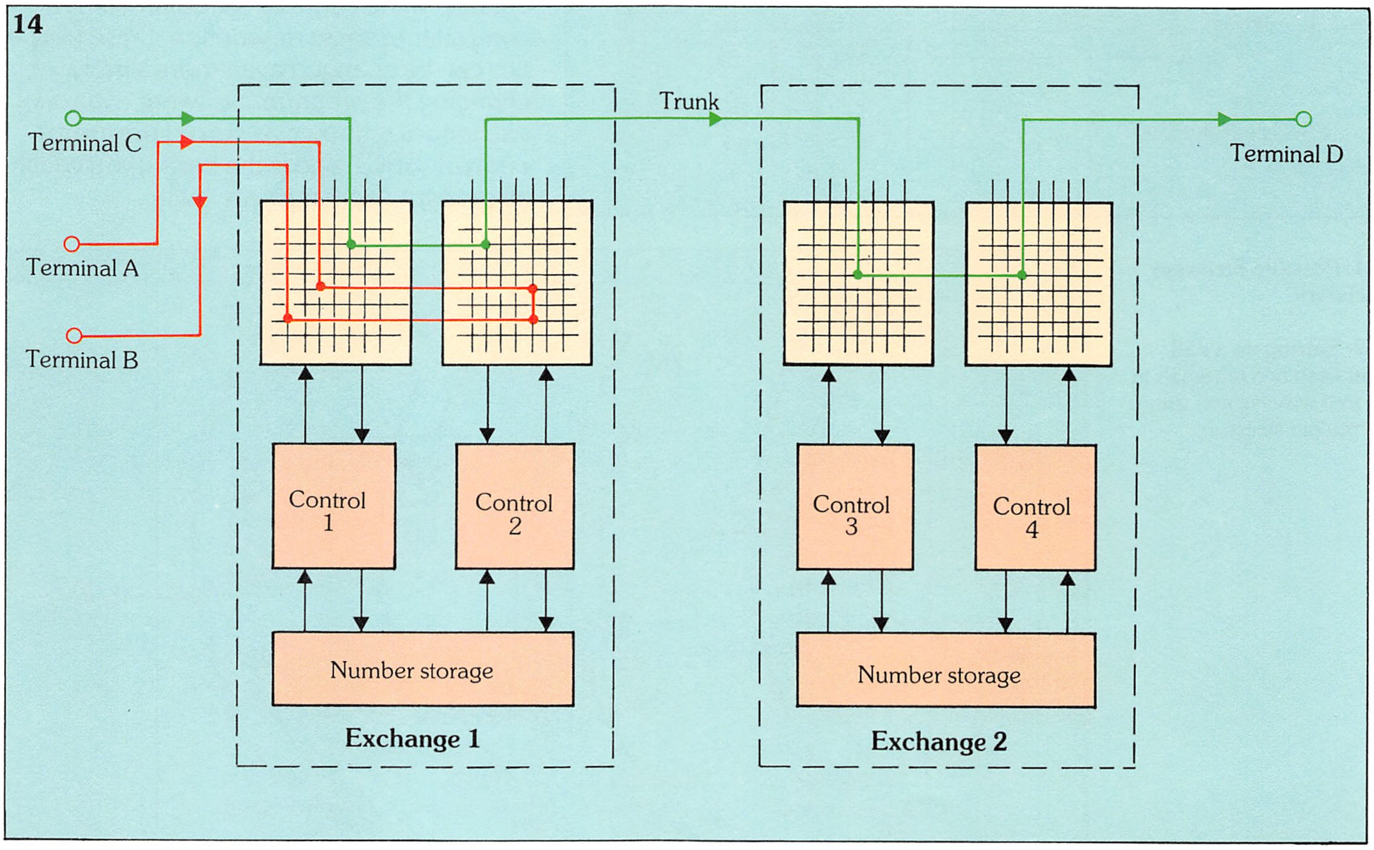


Digital telephone systems and the future

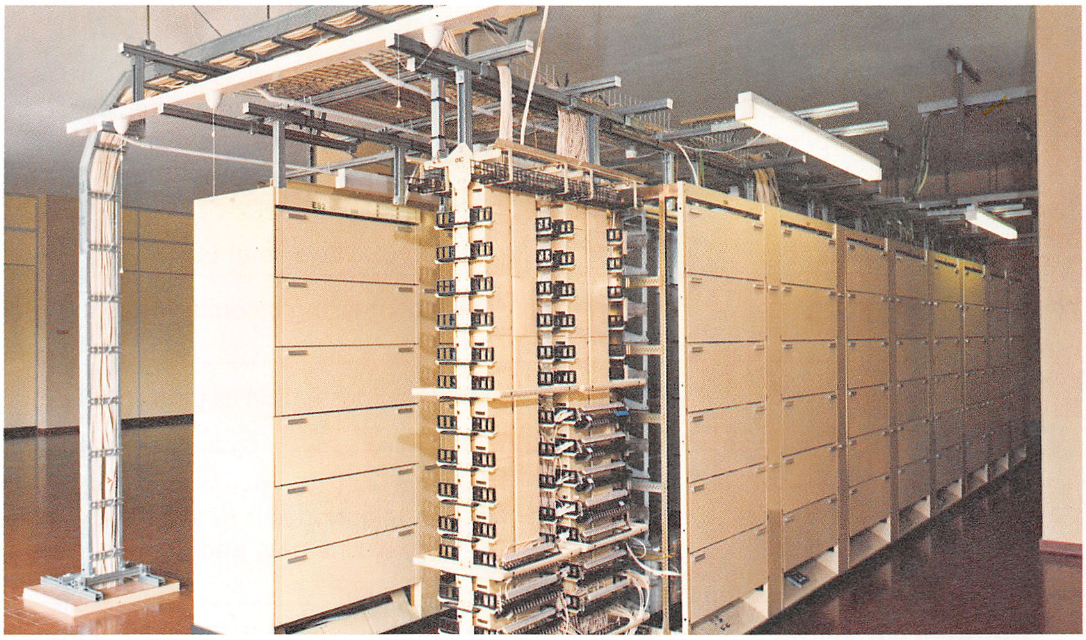
The use of stored program control, which is digitally based, leads us to consider the possibility of digital switching of telephone links. After all, the electromechanical Strowger and crossbar switching mechanisms are large, ungainly, and expensive methods of circuit switching. Literally miles of wires and thousands of moving parts make up a large Strowger exchange, which means that it will be fault-prone; and when



13. Reed relay switch.



14. Switching using crossbar or semi-electronic exchanges.



The Research House/British Telecom

Right: System X exchange at Baynard House.

a fault occurs it takes considerable time to find and repair. Digital switches, on the other hand, are totally electronic and solid state, so faults are less likely to occur, and are much quicker to locate and repair.

The disadvantage of digital switches is that they are not really suited to switching analogue signals in the type of networks which make up a telephone system. They have finite on and off resistances which cause analogue signal loss and leakage currents, and in a network where many thousands of these switches are connected – as in an exchange – these will be of a significant level.

But digital switches *can* be used, with no problems, for digital signals, because loss and leakage currents do not matter where signals with only two levels are concerned.

It is this main fact which has led to the thinking behind the development of a new telephone system within the U.K., known as **System X**. The strategy of System X is that if *all* signals within the telephone system are digital (and that includes local line signals between terminals and local exchanges), then digital switching techniques and stored program control may be used throughout.

A further advantage of digital switching and signalling between terminals and exchanges is that the telephone lines may also be used for high-speed transmission of data, forming an **integrated services digital network** (ISDN). For example, a single telephone speech path, if digital, could transmit data at 64,000 bits per second. This data path may be used for computer-computer data links, viewdata links, electronic mail systems, facsimile transmission etc. The possibility also exists, of using more telephone links for higher data rates.

However, the total ISDN of System X is still a long way off. The task to convert an existing analogue network, the size of the U.K. telephone system, is enormous and is scheduled to be completed in stages.

The first stage, due for completion shortly, is to provide a national digital trunk network. Secondly, all Strowger-based local exchanges and all trunk and tandem exchanges will be replaced with digital exchanges – by the mid-1990s.

Finally, all crossbar, existing semi-electronic exchanges, and loop-disconnect terminals will be replaced with the corresponding digital equipment by the year 2014.

Although this might seem a long time to implement, it should be remembered that with over 6000 exchanges and 20 million terminals, the task is by no means small. The extra facilities and quality of service provided will make the telephone system in the U.K. one of the most advanced communications systems in the world.

Glossary

carrier signal	a signal which is modulated with a message signal, in order that the message signal may be transmitted
crossbar	type of electromechanical exchange used in telephone systems
crosspoint	a switch within a crosspoint array matrix
crosspoint array matrix	an array of switches connecting incoming lines to outgoing lines in an exchange
envelope amplitude modulation	a technique for sending information as patterns of amplitude variations of a carrier signal
four-wire operation	telephone line formed with two twisted pair wires, in which speech transmission between telephones is in two different directions, i.e. the go signal is transmitted in one pair, and the return signal is transmitted in the other pair
frequency division multiplex (FDM)	the allocation of frequency ranges within a transmission medium's overall bandwidth, to transmit many message signals through the single medium
go, return	the speech channels between calling and called terminals, and called and calling terminals, respectively, in a communications link which is not of two-wire operation
group	12 frequency division multiplexed telephone speech channels
hybrid transformer	transformer which allows two-wire and four-wire operating telephone circuits to interface
jumbo group	6 frequency division multiplexed mastergroups (3600) channels
mastergroup	10 frequency division multiplexed supergroups (600 channels)
sideband	bands on either side of the carrier of an amplitude modulated signal, corresponding to the message signal
single sideband suppressed carrier modulation	FDM technique in which only one sideband is transmitted
singing	problem of oscillation occurring in twisted pair telephone lines, if repeater gain is too high
singing margin	loss, which a section of twisted pair transmission line containing two repeaters must have, to prevent singing
Strowger selector	electromechanical rotary switch, which may be used to form cross-point arrays in telephone exchanges
supergroup	5 frequency division multiplexed groups (60 channels)
two-wire operation	single twisted pair telephone lines along which communication is two-way

ELECTRICAL TECHNOLOGY

Two-port networks

Most of the electrical networks that have been studied in previous *Basic Theory Refreshers* have had two terminals, both connected to a voltage generator. Two terminal circuits such as this are known as **one-port networks**, the port representing the two terminals at which a voltage is applied, and into or out of which a current flows.

We have also met some networks with more than two terminals, the transformer, for example, has one pair of terminals connected to the supply, and a second pair connected to the rest of the circuit. These two pairs of terminals form the two ports of a **two-port network**.

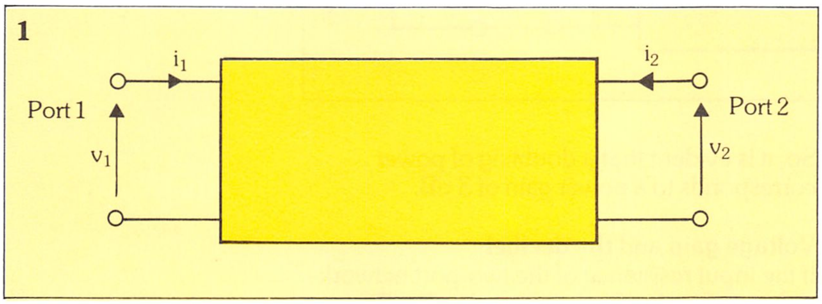
Two-port network models

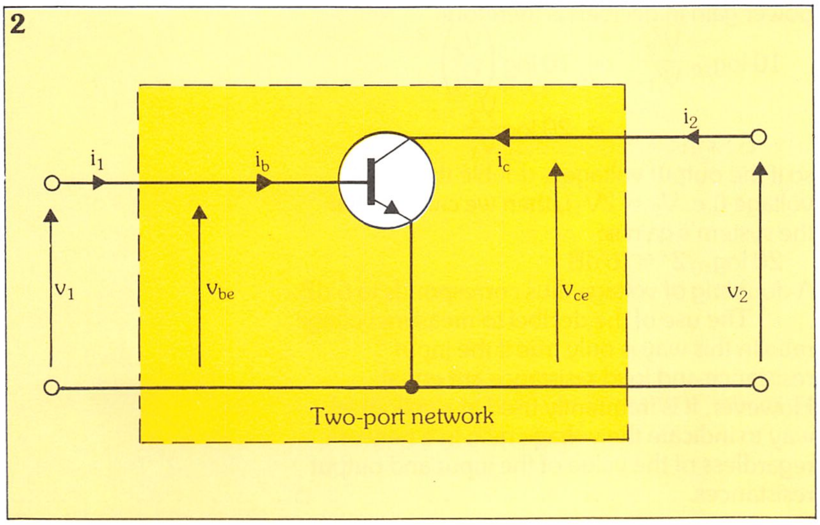
Figure 1 illustrates a model that can be used for a two-port device. Voltage v_1 at port 1 gives rise to a current i_1, while the applied voltage, v_2, at port 2 causes a current i_2 to flow. Alternatively the currents i_1 and i_2 could be thought of as the cause, and the voltages v_1, and v_2 as the effect. In fact, we can find any combination of these four variables and choose two of them as cause and two of them as effect.

The transistor is another common example of a two-port network, if we consider the emitter to be a common terminal shared between input and output. The input port is thus between base and emitter, while the output port is between collector and emitter – as shown in *figure 2*.

1. Circuit model of a two-port device.

2. The transistor as a two-port network.





The relationship between the various currents and voltages in this example are given by the transistors' collector (output) and base (input) characteristics. The output characteristic shows the variation of collector current with collector voltage for a variety of different base currents. The input characteristic shows the relationship between base current and base voltage for different collector voltages. It is not very easy to express these relationships in mathematical form since the currents and voltages are not linearly related to each other.

Admittance parameters

However, if we consider variations of the current and voltage about some operating point, it is possible to define a pair of equations which relate the rms values of the currents and voltages in any two-port device as follows:

$$I_1 = g_{1.1} V_1 + g_{1.2} V_2$$
$$I_2 = g_{2.1} V_1 + g_{2.2} V_2$$

In these equations we assume that the variations of voltage are the cause and those of current, the dependent variable (effect).

We can see that the voltages in the two-port equations are expressed in terms of the currents, so they are known as the **admittance parameters**. The four conductances, $g_{1.1}$, $g_{1.2}$, $g_{2.1}$ and $g_{2.2}$, may be identified with quantities that we can measure.

If port 2 is short circuited, V_2 is forced to zero and so the two equations above become:

$$I_1 = g_{1.1} V_1; \quad I_2 = g_{2.1} V_1$$

So we can see that $g_{1.1}$ is the ratio of input current to input voltage (namely the input conductance), when the output is shorted. The second equation shows us that $g_{2.1}$ is the ratio of output current to input voltage; a transfer conductance (or transconductance) when the output is shorted.

In the same way, short circuiting the input port gives us:

$$I_1 = g_{1.2} V_2; \quad I_2 = g_{2.2} V_2$$

These equations indicate that $g_{1.2}$ is the transconductance between the output port and input port when the input is short circuited; and that $g_{2.2}$ is the output conductance at port 2 under the same conditions.

Voltage and current gain

Now let's see how these network models can

be used in a complete circuit. Consider an amplifier; we can connect the input to a voltage generator, which can be modelled by a voltage source, V_S, in series with a resistor R_S. The output will be connected to a load resistor R_L as shown in *figure 3*.

The current and voltage at the source are related by:

$$V_S = R_S I_1 + V_1$$

while across the load we have:

$$V_2 = R_L I_L$$

Finally, because of the convention that we have adopted for the directions of currents:

$$I_2 = -I_L$$

These three expressions enable the voltage gain k_v ($= V_2/V_1$) or the current gain, k_i ($= I_2/I_1$), of the two port network to be found:

$$k_v = -\frac{g_{1.2}}{g_{2.2} + 1/R_L}$$

$$k_i = \frac{g_{2.1}}{g_{1.1} + (g_{1.1}\,g_{2.2} - g_{1.2}\,g_{2.1})\,R_L}$$

Power gain and the decibel

We can use the ideas above to determine the power into and out of a two-port network. The input power is $P_1 = V_1 I_1$, while the output power $P_2 = V_2 I_2$.

So the power gain k_p of our amplifier, is:

$$k_p = \frac{V_2 I_2}{V_1 I_1}$$

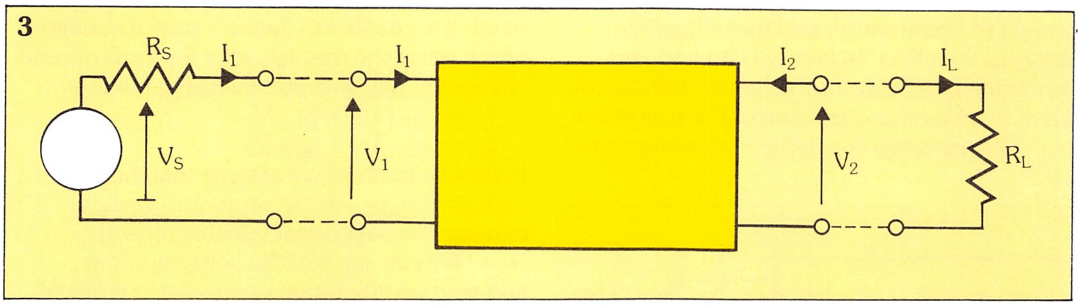
Power gain is often conveniently expressed in decibels. The power gain of a network in decibels is thus given by:

$$10 \log_{10} \frac{P_2}{P_1}$$

The decibel is a convenient unit to use in such cases as audio amplifiers, because the human ear is sensitive to logarithmic increases in power, and not to equal increases in the power itself.

If we consider a system in which the output power is twice the input power, the power gain in decibels (dB) is:

$$10 \log_{10} \frac{2P_1}{P_1} = 10 \log_{10} 2 = 10 \times 0.3$$
$$= 3\text{ dB}$$



3. Determining the voltage and current gain in a two-port network.

Input and output resistance

We have seen that when the output port is connected to the load resistance, R_L, the input conductance, $G_{in} = I_1/V_1$, is given by:

$$G_{in} = g_{1.1} - \frac{g_{1.2}\,g_{2.1}}{g_{2.2} + 1/R_L}$$

so you can see that the two-port parameters allow the calculation of all the important properties of a network. This is crucial when dealing with amplifiers.

Frequency dependent two-port parameters

So far we have concentrated on a two-port network that was defined by its conductances. However, this implies that the network is unaffected by frequency. The two-port parameters in most small signal networks *are* dependent on frequency, as there may be reactances incorporated in some or all of the element values. This means that we should specify the network by four admittances, rather than the simple conductances.

So, it is evident that a doubling of power corresponds to a power gain of 3 dB.

Voltage gain and the decibel

If the input resistance of the two-port network of *figure 4* is also R_L then $P_1 = V_1^2/R_L$. The power gain in decibels is therefore:

$$10 \log_{10} \frac{V_2^2}{V_1^2} = 10 \log \left(\frac{V_2}{V_1}\right)^2$$
$$= 20 \log \frac{V_2}{V_1}$$

so if the output voltage is double the input voltage (i.e. $V_2 = 2V_1$), then we can see that the system's gain is:

$$20 \log_{10} 2 = 6\text{ dB}$$

A doubling of voltage thus corresponds to 6 dB.

The use of the decibel to measure voltage ratio in this way is only true if the input resistance and load resistance are equal. However, it is frequently used in a colloquial way to indicate the voltage ratio in a network, regardless of the value of the input and output resistances. □

DIGITAL ELECTRONICS

26 How digital systems function-1

How many bits?

We know that digital systems handle information by representing it as bits: these bits are processed by simple gate circuits and can be stored in many different ways (*figure 1*). Gates and storage units can be linked together in any number of combinations to handle nearly any kind of information processing task. As we have seen, certain general circuit patterns are useful as **building blocks**: flip flops; registers; counters; and multiplexers, for example. Now we need to look at the way that these building blocks are combined to perform various functions in different digital systems.

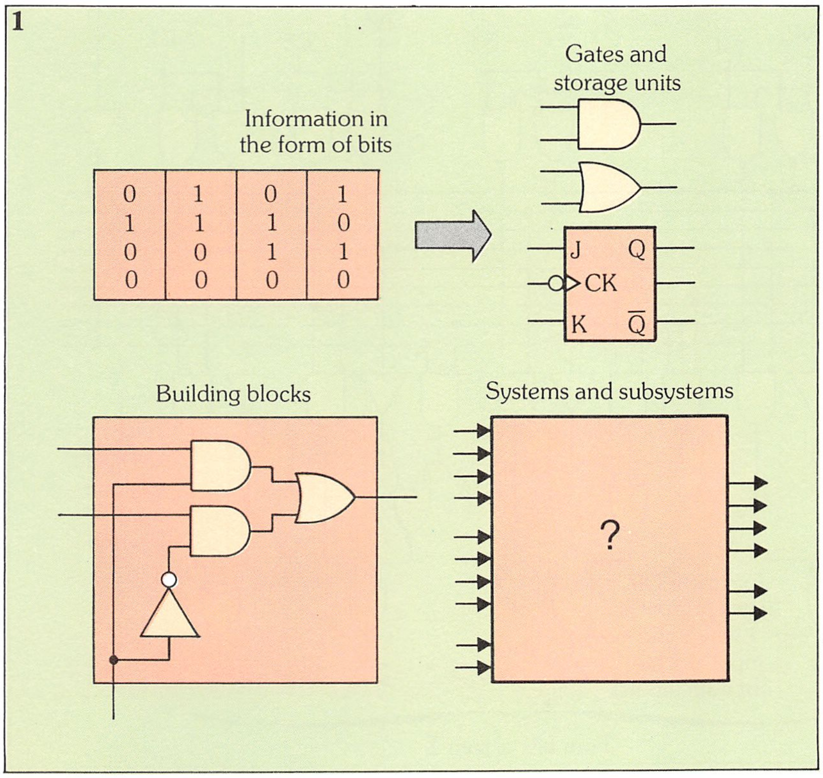
This is a vast problem, for as we have seen, there are many different applications of digital electronics: from our simple calculator example to large computer systems. Obviously, we need to simplify our examination of digital systems, and study their common aspects. In this and the following chapter we will be looking at three basic decisions that the digital electronics engineer must make during the design process: how many bits have to be worked on at once; whether to use hard-wired or variable programming; and should dedicated or multipurpose hardware be used. These points will be considered one at a time.

The first general question is: how many bits do we need to work on at once? We know that information leaves and enters digital systems in the form of groups of bits. However, once inside the system these groups can either be broken down into smaller groups and processed separately, or perhaps combined into larger groups and processed together. This breaking down and recombination of groups of bits may occur at several different stages in a system, depending on what the designer decides works best.

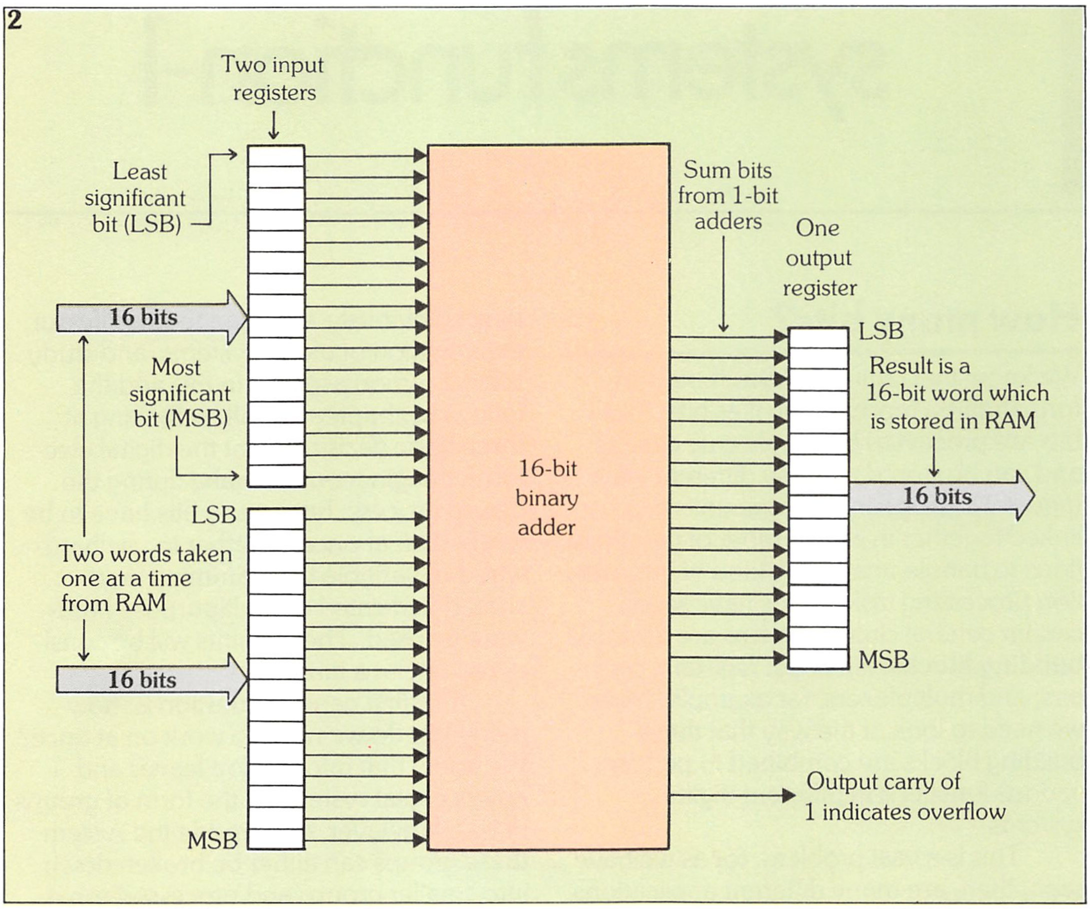
In the most basic terms, it is a choice between parallel or serial processing to perform the same function. The example of binary addition that was looked at in *Basic Computer Science 3* will serve to illustrate the alternatives involved.

First of all, let's look at **full parallel addition**, i.e. all the bits are added at once, without being broken into smaller groups. *Figure 2* illustrates the process involved when two 16-bit numbers are added in full parallel fashion. This method may be used, for example, in the ALU of a 16-bit computer. Two words are routed from RAM (not shown), placed in two parallel 16-bit input registers and then routed to the 16-bit adder. The resulting 16-bit sum word is put into an output register, from which it is later routed into the main memory.

This example shows that all the bits of

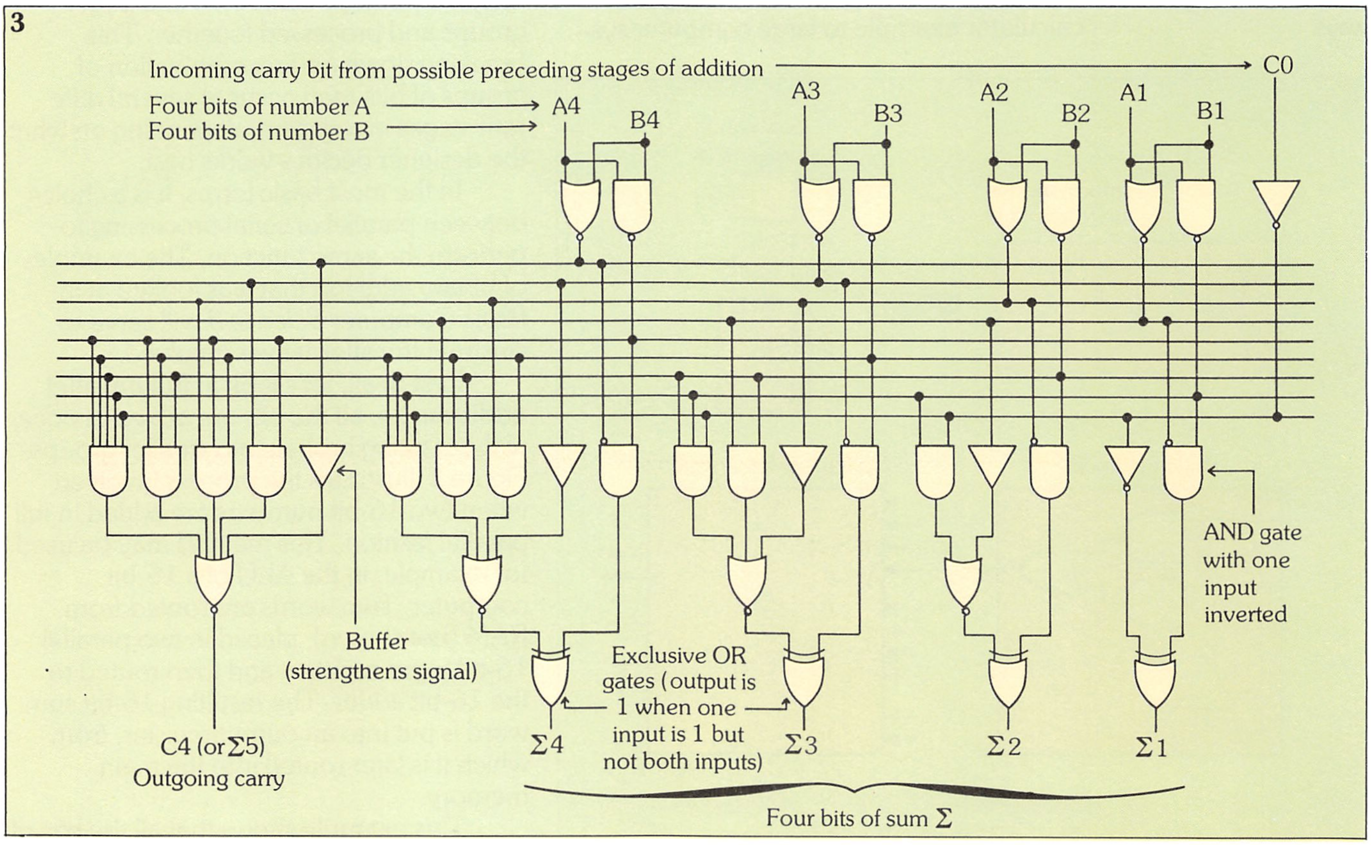


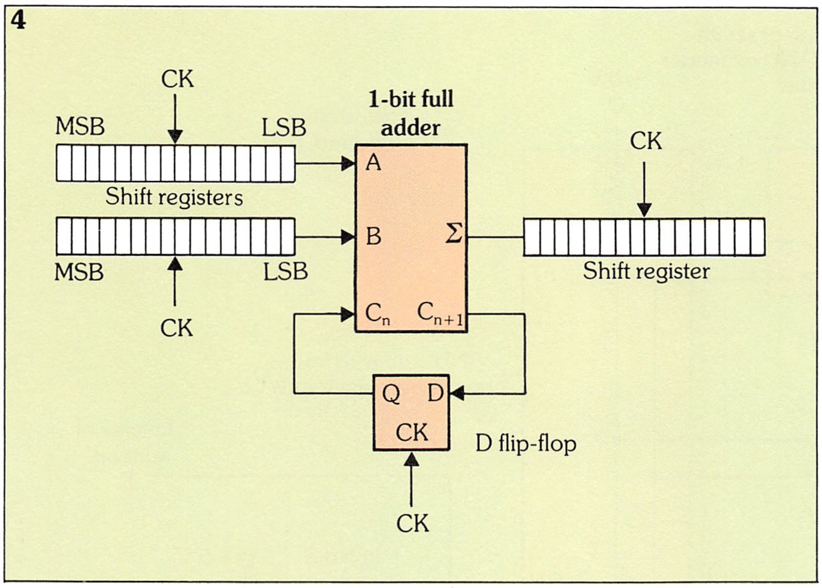
1. Digital information can be processed and stored in a number of ways.



2. Full parallel addition.

3. 4-bit carry-look-ahead adder.





tion of the sum and the performance of any carrying without the need to wait for carry bits to pass (ripple) between 1-bit adders. However, using this method, the number of gates rapidly increases as the number of bits to be handled increases. This, of course, affects the final cost of the circuit.

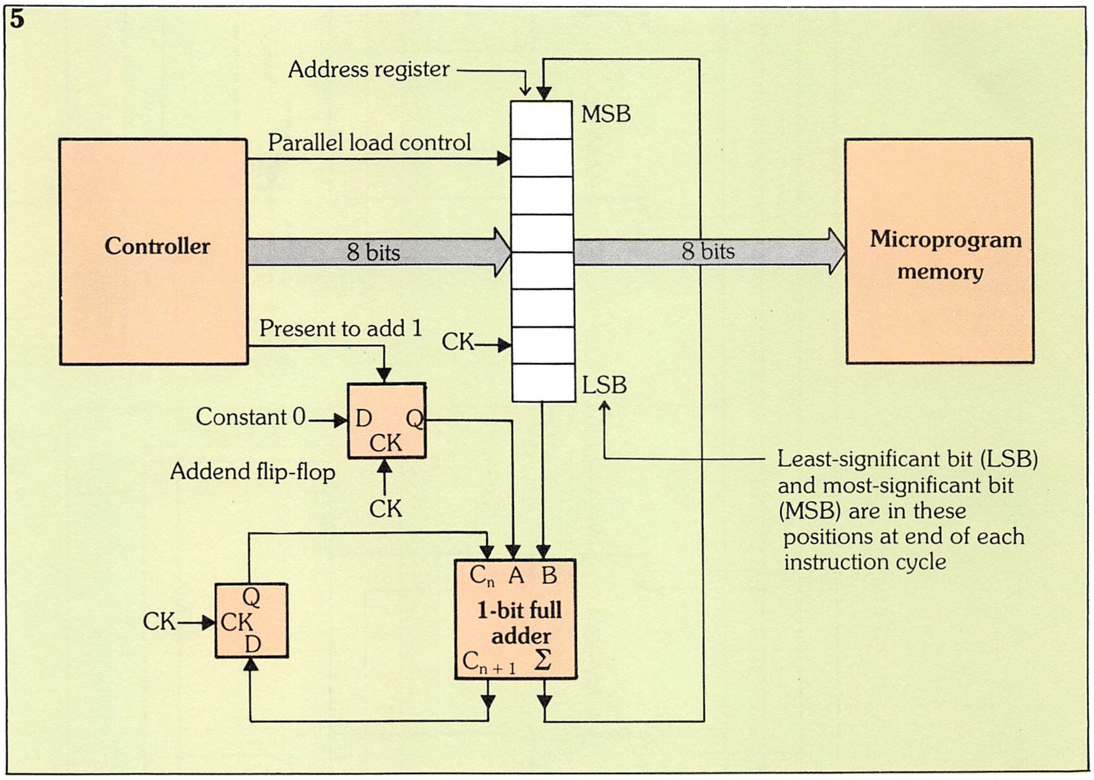
Full serial binary addition

The addition of binary numbers in a completely serial way, means adding the bits of two numbers one at a time, using the same 1-bit full-adder for all the different bits. *Figure 4* shows the addition of two 16-bit numbers.

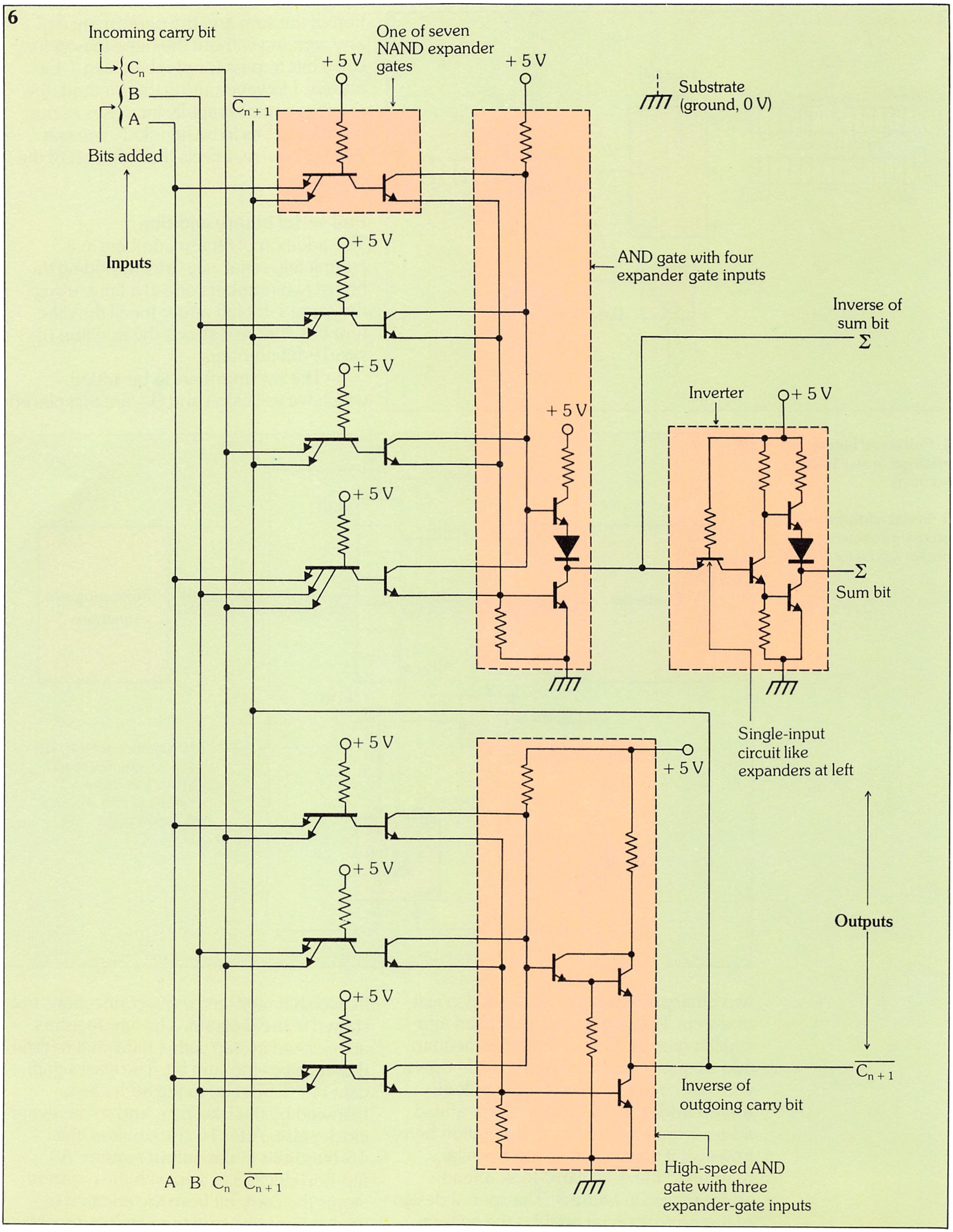
The two numbers to be added – which we will call A and B – are first placed in two shift registers, using connections not shown in the diagram. The two registers are clocked in step with a third shift register that receives the sum bit. The least significant bits (LSBs) go to the adder first, followed by the next pair, and so on, every clock pulse. After 16 clock pulses, the 16-bit sum is in the output register. A flip-flop clocked in step with the registers stores the carry bit from the addition of each pair of bits, and then passes it to the

4. Full serial binary addition of two 16-bit numbers.

5. Serial addition in the program counter of a pocket calculator.

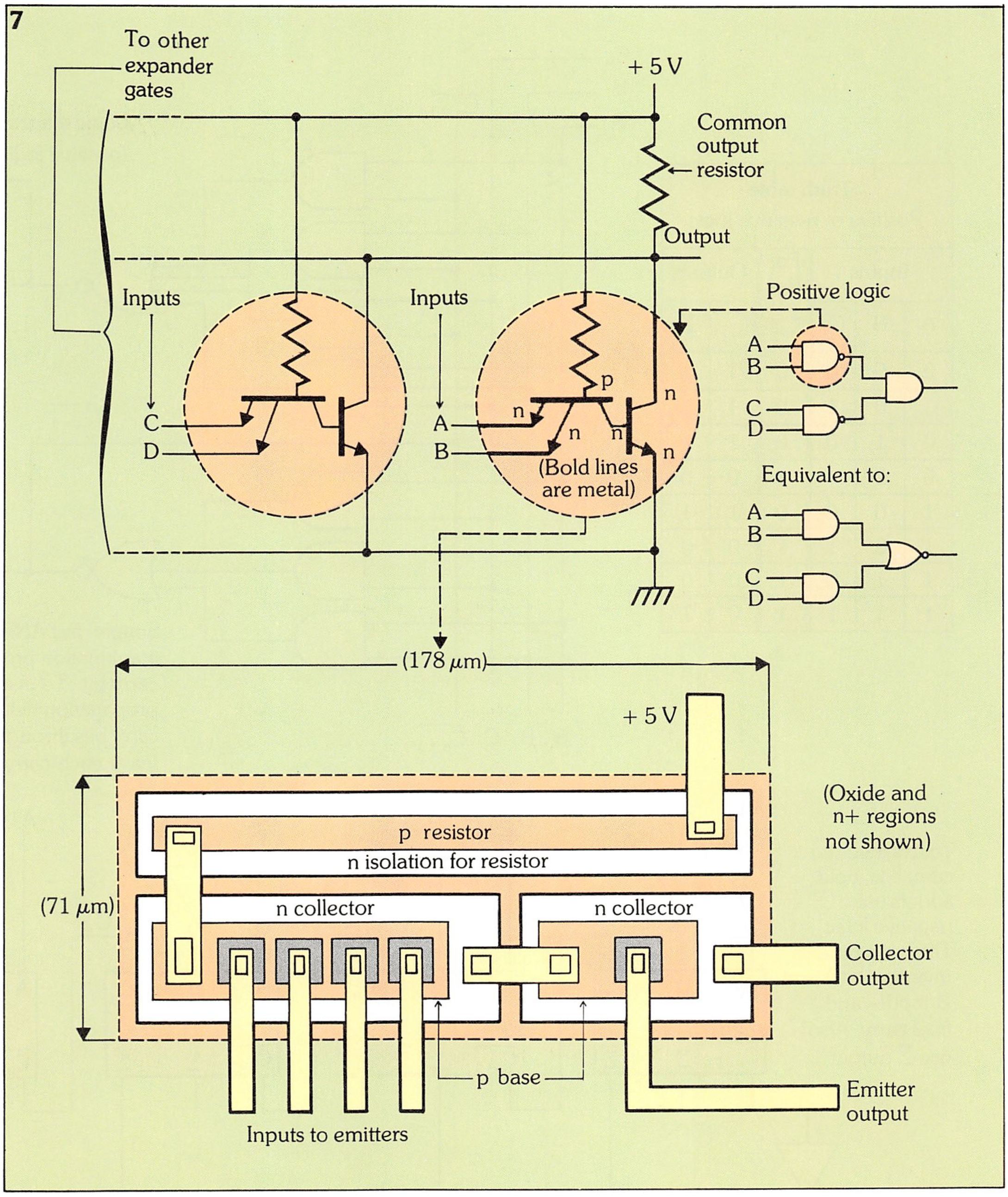


two binary numbers can be added in just one step. The words are not broken into smaller groups. Since computers need to add as quickly as possible, the ALU uses a very fast circuit known as a **carry-look-ahead adder**. A 16-bit carry-look-ahead adder is too complex for examination here, however it would function in a similar manner to the 4-bit carry-look-ahead adder shown in *figure 3*. The special design of this type of circuit enables the genera-


6
Incoming carry bit
C_n
B
A
Bits added
Inputs
$\overline{C_{n+1}}$
+5 V
One of seven NAND expander gates
AND gate with four expander gate inputs
Substrate (ground, 0 V)
Inverse of sum bit
Σ
Inverter
Sum bit
Single-input circuit like expanders at left
Outputs
Inverse of outgoing carry bit
High-speed AND gate with three expander-gate inputs
A B C_n $\overline{C_{n+1}}$

6. Schematic diagram of a 1-bit adder – the 7480.

7. Detail of one of the seven NAND expander gates of the 1-bit adder in *figure 6*.



adder, to be added in with the next pair.

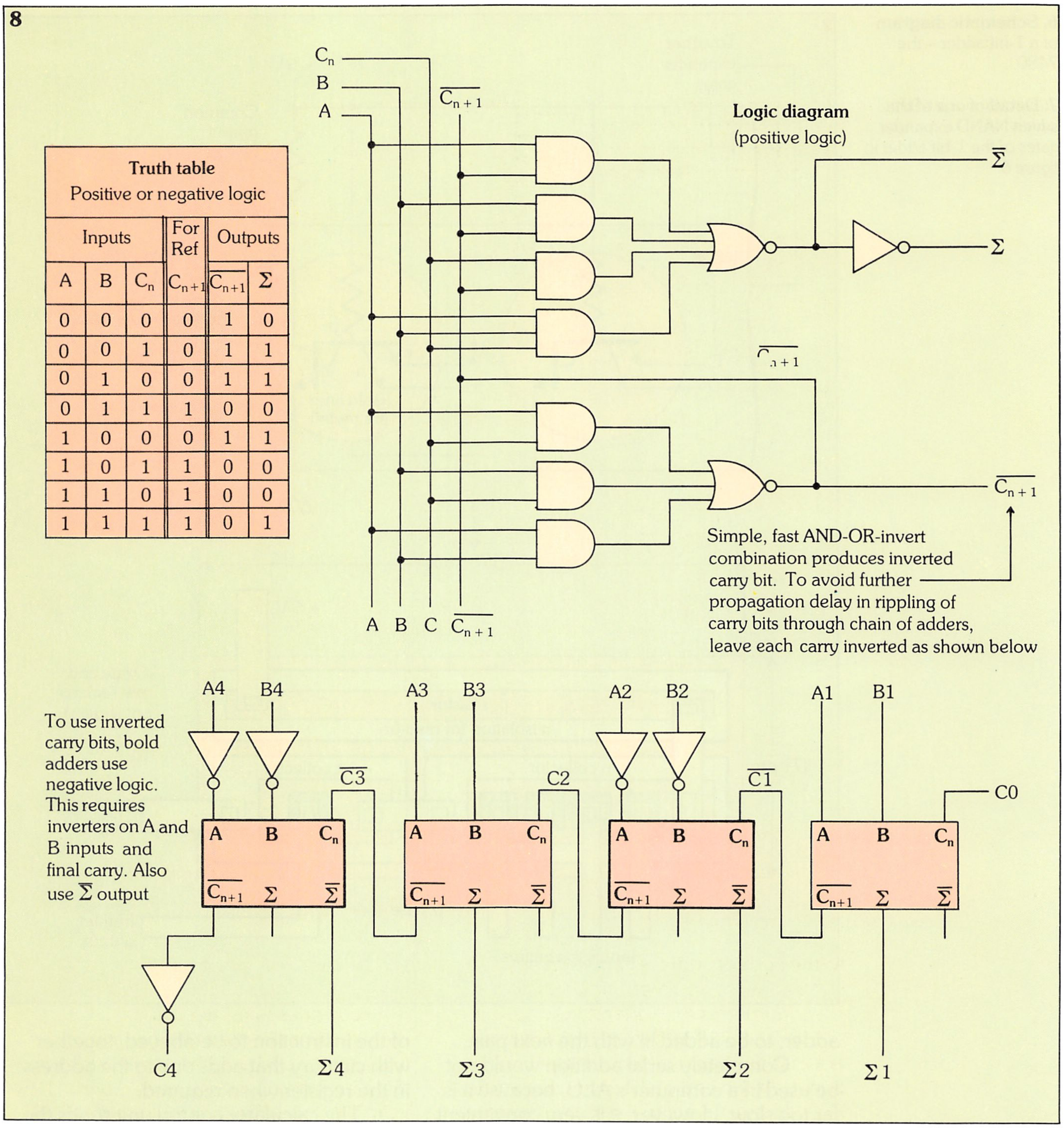
Completely serial addition would not be used in a computer's ALU, because it is far too slow. However, it is very convenient when cost and space saving is required but high speeds are not. The program counter in our calculator example provides a good illustration of the use of serial addition.

Serial addition in the program counter

The instructions in a pocket calculator are permanently memorised in the microprogram ROM. The program counter consists of a register that holds the memory address of the instruction to be obeyed, together with circuitry that adds one to the address in the register when required.

The calculator control unit sends the contents of the program counter to the microprogram memory, which answers by supplying the instruction to be followed. The program counter is increased by one during the instruction cycle, so that it contains the address of the next instruction to be carried out, enabling the control unit to proceed step by step along the program routine. *Figure 5* illustrates how the calculator program counter uses full serial addi-

8



Truth table
Positive or negative logic

Inputs			For Ref	Outputs	
A	B	C_n	C_{n+1}	$\overline{C_{n+1}}$	Σ
0	0	0	0	1	0
0	0	1	0	1	1
0	1	0	0	1	1
0	1	1	1	0	0
1	0	0	0	1	1
1	0	1	1	0	0
1	1	0	1	0	0
1	1	1	1	0	1

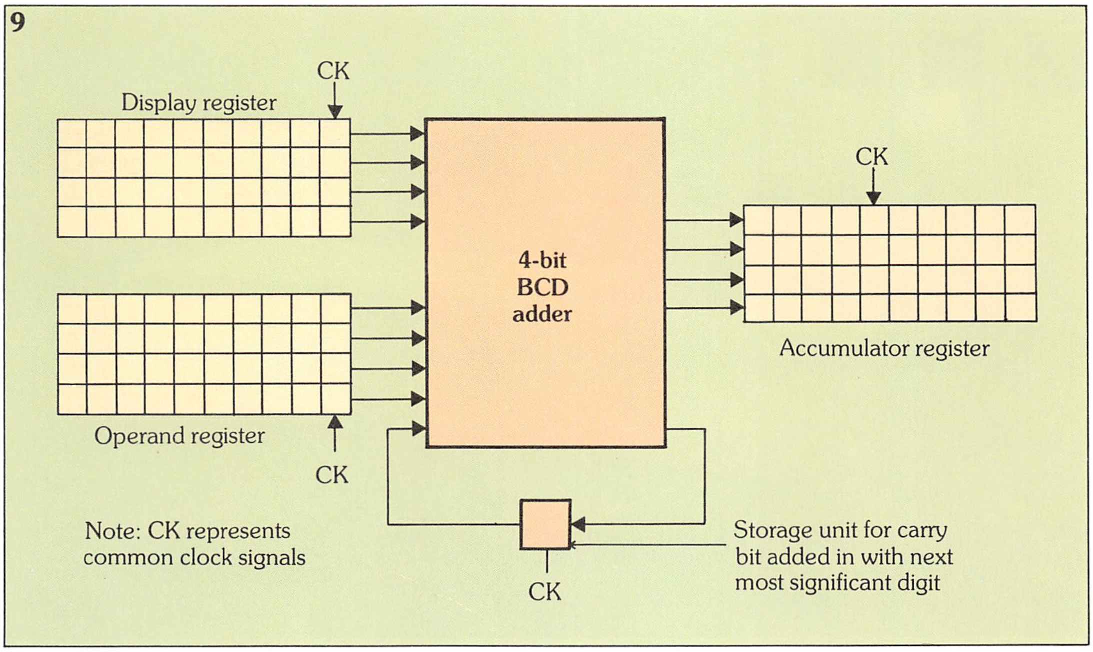
tion for this purpose.

Comparing the programme counter with the full serial.adder in *figure 4*, you'll note that the principle is the same, but the program counter's address provides one input number B and receives the sum. This is an 8-bit shift register with parallel inputs and outputs. For this reason we'll assume that our calculator's addresses consist of eight bits each.

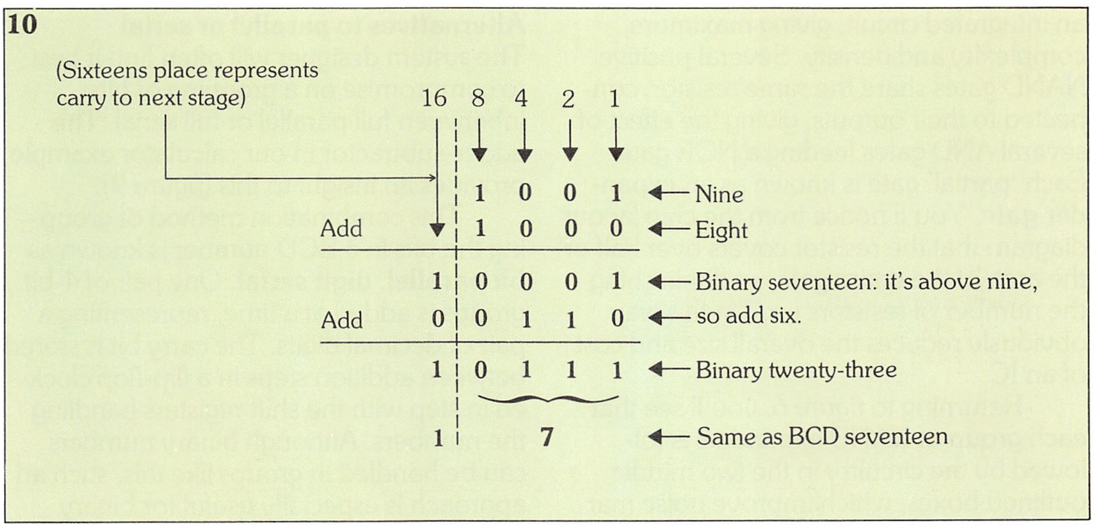
The other input number, A, is provided by the **addend flip-flop**. This unit has a constant 0 at its data input, but can be preset to 1 by the controller at the beginning of each instruction cycle.

The address register and both flip-flops are clocked eight times during each instruction cycle. If nothing is to be added

8. Logic diagram and truth table for the circuit in *figure 6*.



9. Bit-parallel, digit serial combination method of grouping the bits used in the adder-subtractor of a pocket calculator.



10. BCD addition of the numbers 8 and 9.

to the current address (if the current instruction is to be repeated) the controller does not preset the addend flip-flop. So the constant 0 is added to each bit of the address as it recirculates through the adder. At the end of the instruction cycle, the address returns to the correct position, during which time the microprogram memory refers to it and fetches the instruction at that address.

To add 1 to the address, the controller presets the addend flip-flop to 1 at the beginning of each instruction cycle. This 1 is added to the LSB of the address during the first clock cycle. During the remaining seven cycles the addend flip-flop presents the constant 0 to be added to the other seven bits of the old address.

How complex is a 1-bit adder?

The difference in complexity and size between a 1 and a 16-bit adder may not seem important until it is remembered how building blocks like these are made. So that we don't forget about the inner workings of the devices that are being discussed, look at *figure 6*. Here, a schematic diagram of a 1-bit adder, the 7480, has been taken from the manufacturer's data book with a few amendments.

There are seven NAND expander gates in this circuit, and a close-up of one of these is shown in *figure 7*. This particular type of circuit is designed to be used inside

an integrated circuit, giving maximum complexity and density. Several positive NAND gates share the same resistor, connected to their outputs, giving the effect of several AND gates feeding a NOR gate. Each 'partial' gate is known as an **expander gate**. You'll notice from the chip layout diagram that the resistor covers over half of the area of the entire gate, so diminishing the number of resistors used in this way obviously reduces the overall size and cost of an IC.

Returning to *figure 6*, you'll see that each group of NAND expanders is followed by the circuitry in the two middle outlined boxes, which improve noise margin and fan-out. Together, these circuits act as a group of NAND gates feeding an AND gate. We know from *figure 7* that this is equivalent to a group of AND gates feeding a NOR gate – known as an **AND-OR-invert** combination. You'll also notice that the inverter in the outlined box on the right of *figure 6* consists of one single input TTL expander circuit followed by circuitry similar to that found in the upper middle outlined box.

The logic diagram and truth table for the circuit in *figure 6* is shown in *figure 8*, assuming that positive logic is used. It must be obvious now how complex even a simple 1-bit adder can be, so you'll realise why full parallel addition should be avoided whenever possible.

Alternatives to parallel or serial

The system designer will often find it best to compromise on a grouping of bits inbetween full parallel or full serial. The adder-subtractor in our calculator example provides an insight to this (*figure 9*).

This combination method of grouping the bits in a BCD number is known as **bit parallel, digit serial**. One pair of 4-bit groups is added at a time, representing a pair of decimal digits. The carry bit is stored between addition steps in a flip-flop clocked in step with the shift registers handling the numbers. Although binary numbers can be handled in groups like this, such an approach is especially useful for binary coded decimal numbers, which the computer uses. We looked at BCD addition in *Basic Computer Science 3*, but let's just recap.

BCD addition adds digits as if they were ordinary 4-bit binary numbers and then adds six if the sum is greater than nine. *Figure 10* shows an example of how this works by adding the numbers 8 and 9. As you can see, the result is 10001 – binary seventeen. As this is above nine we add six, giving us binary twenty three – which is 10111. This result is then treated as a carry of 1 and a BCD sum of 0111 – giving us seventeen in BCD form.

(continued in part 30)

Left: A Series computer (Photo: Burroughs).